THE SOUND ENGINEERING MAGAZINE

OCTOBER 1983
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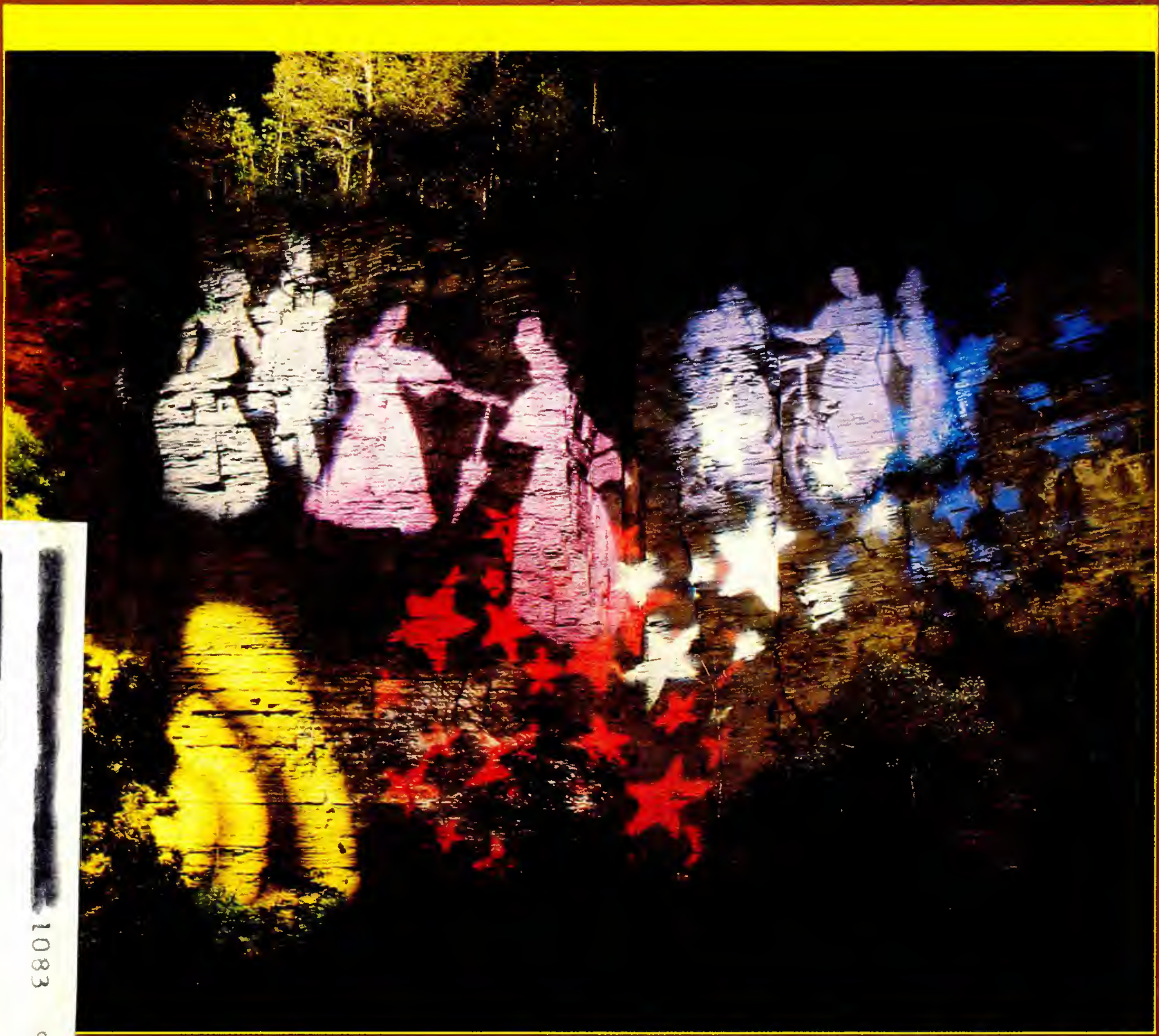

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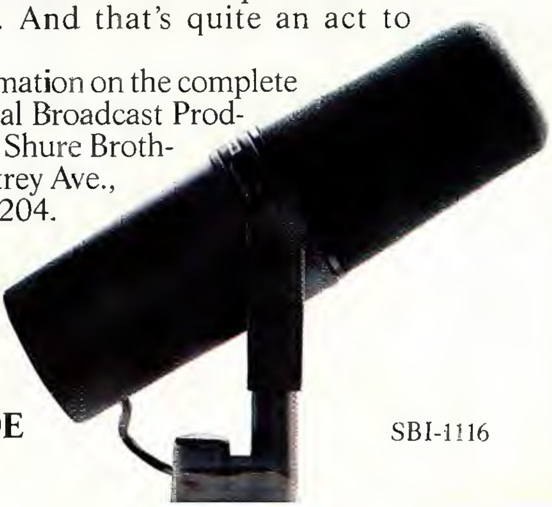

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FEATURES

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Letters

LOST IN SPACE

TO THE EDITOR:

I am a regular follower of John Eargle's column on Sound Reinforcement, and have suddenly found myself in a dilemma from which I cannot rest without an explanation. In the September issue, Eargle's column is entitled, "Microphones in Sound Reinforcement, Part 3." Have I been in the twilight zone for parts 1 and 2, or is this some kind of cruel joke? Earth to **db**...please respond.

JANET SORENSON
Houston, Texas

db replies:

Boy it's tough to fool **db** *readers! Skip two measly columns and they jump down your throat. No, you're not in the land of Rod Serling. For the full story behind this unfortunate incidence (makes it sound like international politics, doesn't it?), see John Eargle's column on page 19.*

WE'VE COME NOT TO PRAISE

TO THE EDITOR:

After all the published articles and test reports praising or castigating the development and acceptance of the CD, I found Michael Tapes' article on CDs (for once) refreshing. I think Compact Disc technology is a great concept that still needs a bit of debugging in the area of recording techniques and perhaps in the electronics themselves. The product holds much promise; the *lack of noise* sounds great, but to say that the music sounds great is ludicrous. However, I haven't closed my mind. It's a great idea—to be listened to at a later date. Thanks, Michael.

MAX FOLEY
Max's Recording
Toledo, Ohio

Index of Advertisers

* *

COMING NEXT MONTH

- In November, **db** celebrates its 16th birthday! To mark the occasion, we'll be featuring a special insert commemorating our first issue, complete with ads and "new products" from 1967. We'll also be featuring articles updating ones that appeared in that first issue. In addition, our new feature "In My Opinion" returns, along with articles on Bearsville Studio, audio/musical designs for animated shows, and more. Of course, our regular columnists and departments will also be on hand. All this—and more—in November's **db—The Sound Engineering Magazine**.

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WE WANT MORE

TO THE EDITOR:

I found Greg Hanks' article on modifying the recording process for film production very informative and am looking forward to Part II. Knowing little about audio as it relates to film production, I find this type of article particularly interesting and encourage **db** to publish more of this kind of information. I would also like to see more computer-related articles, as computers are being used more and more in the audio world. Thanks for a magazine that I still learn a great deal from.

GENE BAXTER
Philadelphia, PA

db replies:

Your letter comes at just the right time for us to tell you and all our readers that what you want is happening! We will be having many more articles about audio in video and on film. We also will have a regular column on computers in the pro audio world. It's already in the works.

In My Opinion

Patents, Copyrights and "Pirating"

This month db inaugurates a new feature. "In My Opinion" is intended (surprisingly enough) to be a forum for ideas on the state of the audio industry. If you've got something to say that's more than a letter but not quite an article, we've now got you covered.

Kicking off the proceedings this month is Norman Crowhurst, a former db columnist.

So, don't forget. If you're happy, sad, confused, mad, etc. about something audio, write it down, send it in, and let all of us know what's on your mind.

• People, both lay and professional, seem confused by the controversy over this subject. It's being referred to the Supreme Court, but is it really a legal problem? Howard Russell, in a letter to the editor in April's **db**, suggested that it was a moral problem. I suggest the problem is really one of perspective.

More than half a century ago I came into electronics which was then a rapidly advancing industry (and you'll say we hadn't seen anything then!). Companies had scores of attorneys working up patent applications on every advancement the company made, to protect their "inventions."

Patents and copyrights were introduced at a time when not many inventions were being made—compared with today, that is. But when I started, the electronics field was already advancing at a pretty good clip. It didn't take young engineers long to realize that by the time most new inventions got patented—a process that could take from one to five years—we were already moving on to something that made that invention obsolete.

So, rather than following the big companies' example, we didn't patent every new thing that looked to be patentable. Of course, the question, "Can the idea or development be patented, anyway?", was something that patent attorneys and patent office examiners have always kicked around —and made a pretty handsome living doing it. But we saw another criterion for deciding whether or not to apply: How central to possible future developments does this new device seem to be?

This criterion caused us to reject a considerable portion of the items for which tradition would have had us apply for patents as not worth the time, effort or expenditure. If the patent was likely to be obsolete by the time it issued, if not before, then why apply for it? Before it issued someone might copy it, true, but we'd have to wait on its issuance to sue for infringement, by which time we'd probably have fresher eggs to fry!

Look at it another way. The whole purpose of patent law is to protect an inventor's investment. He/she spends, or his company spends, a lot of resources developing this invention. Getting a patent is intended to prevent some "rip-off" artist from copying an invention and making it cheaper, thus robbing the true inventor of the rewards for the labor and resources put into developing it, unless royalties are paid that fulfill that purpose.

Okay, so now suppose the expected useful life of an invention is, at the outside, two years. The original reason for applying for a patent is no longer quite valid, although I'm sure someone will argue with me about that! Really good "copy artists" out there may get into production to steal a little bit of the market before it's obsolete. It certainly isn't going to be worth protecting for the relatively long time that's automatic when a patent is granted.

The real time when protection *could* help is before the patent issues, and the patent doesn't really give protection until it issues. It's true the words "Patent Applied for" on a product can be a bit of a deterrent. But certainly not as much of a deterrent as it was in the days when the patent had considerable promised "life" ahead of it. So my company then adopted the following policy.

First, we never resorted to stealing other people's ideas. That would be a waste of time: they were already in production, and by the time we geared up, the thing could well be obsolete. The better way was to be first with the new ideas and inventions, so we could exploit them before moving on to newer things. We realized that using patents to protect an investment of time and resources in producing inventions had become somewhat out of date. There are exceptions to this, of course, but the days when companies printed 50 to 100 patent numbers on their products are, or ought to be, gone. That practice no longer makes any sense. The protection it affords isn't worth what it costs.

The same is largely true about copyright. Any author will confirm this. One book in a hundred may go on selling through decades. The other 99 sell

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well for the first few months, maybe even a year, then taper off as reader interest turns to something newer. Isn't the same true for records—maybe even more so?

This whole question of copyrighting is probably sparked by the enormous amounts of money that have hitherto been possible from such things as a succession of gold and platinum records. This has run into millions, only because the protection afforded by copyright has made rather inordinately large royalties possible. So when someone records a "hit," everyone associated with it makes a killing which they want to protect with copyright law.

When you think about it, you'll realize that this is being a little unrealistic. Could not the argument we applied to the use of patents and patent law, half a century ago, equally apply here? People always buy what's available. If someone with a "hit" record has a good distribution network and floods the market quickly, thus making what would still be a tidy sum from a relatively narrow markup, the "copy artists" wouldn't be able to get started in time to make any money off that item in most instances.

All right, let's say the really big question is movies and video tapes. People in that industry should resign themselves to the fact that the "gravy train" days are over. Quite incidentally, recognizing that as a fact of life may well prove to be an incentive to producing quality entertainment once again, instead of the garbage that so-called entrepreneurs have been struggling to make a fast buck with lately.

Obviously, whichever field we consider, the original producer of something is in the best position to make large quantities of it before anyone else has time to copy it. That means he can set the price low enough, and still make a profit, effectively keeping the copy artists from making any money from their relatively smaller runs of any individual item. Isn't that the logical way of striking a balance in this situation?

Go back to inventions for a moment. Even when patent protection was more valid than it is today, many companies made a more or less regular practice of "stealing" inventions, regarding the cost of being sued for infringement as a cost of doing business. Howard Russell may regard that as a moral question. But in reality it is a fact of life. There is nothing criminal about it: all it leads to is civil suits. If the court ruled it was infringement, they paid! Big deal!

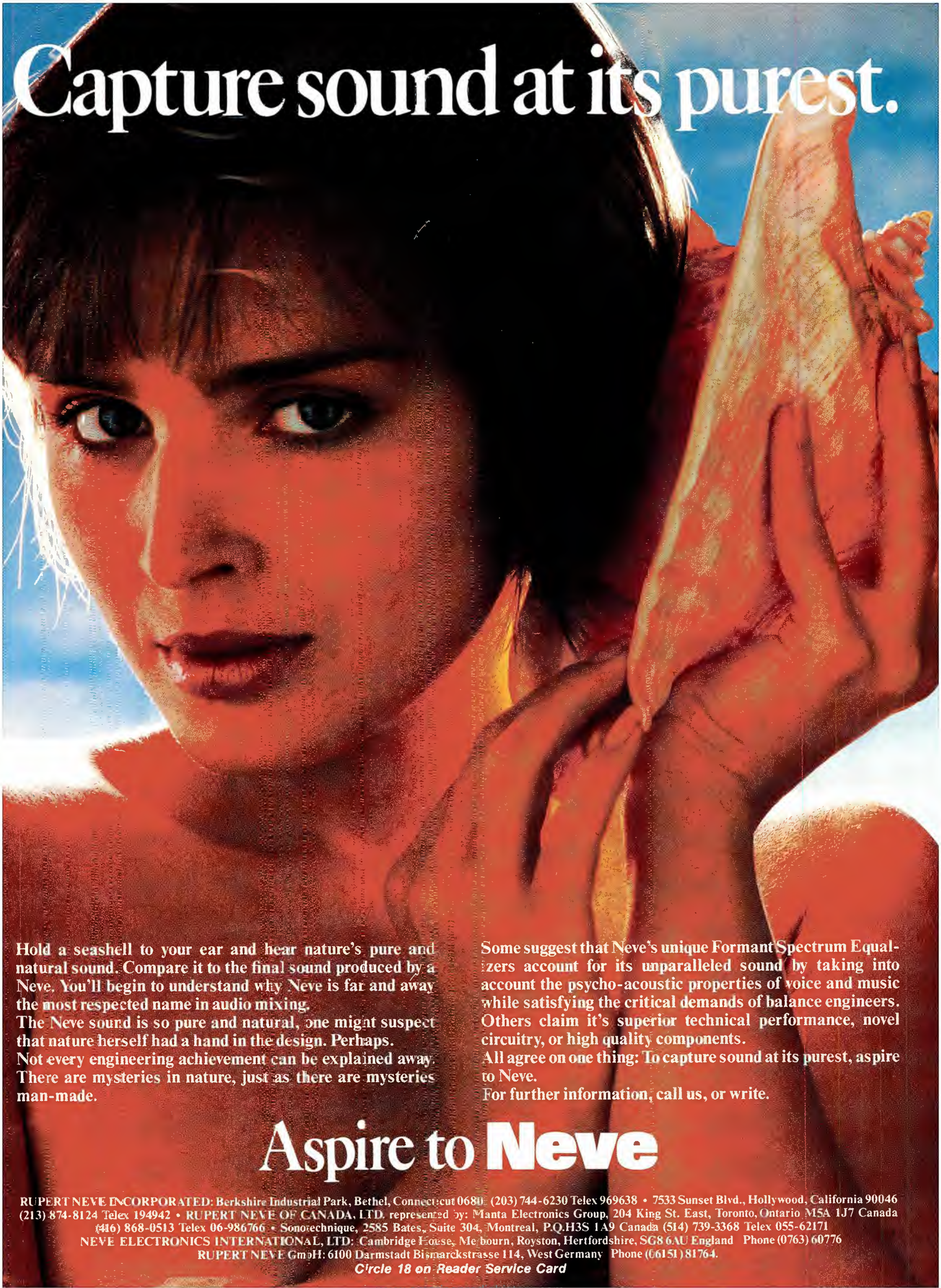
Maybe there's an argument for a reverse psychology. As an author, I've always believed that imitation is a form of flattery (sincere or not). So all I ask from someone who wants to reprint something I wrote is to give me credit for having written it first. That way, if someone wants to read the original material, he/she would know what to ask for. Isn't that fair?

Now, if copying does become a federal crime, who's going to put a notice on whatever it is he copies that says, in essence, "We stole this from ..."? But if on the other hand it's legal to copy, then it would be logical to define a different crime: that is, to copy something *without* acknowledging where it came from. Doesn't that make a lot more sense, and avoid all the conflicts presently raging?

It seems perfectly logical to me. If someone wants to say Crowhurst says..., and then quotes me—thank you for the exposure! He's not stealing, and he may benefit me as well as himself by doing that. On the other hand, if he takes my material and pretends it's his, I would call that stealing!

Yet it is true that a lot of work goes into mastering, working up ideas, and all of that, which the copy artist doesn't have to support. But isn't it conceivable that this copyright and patent protection has encouraged what has really become an extortion racket for that kind of work? It seems to me that what is happening is that the greatly simplified copying processes have brought about a natural way of restoring balance to a situation that has gotten a little "out of hand" in a different direction.

At least, that's my contention. What do you have to say? ■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KEN POHLMANN

Theory & Practice

The Secret of Flux Reversal

• Right off the bat, I can unequivocally state that there wasn't any money involved—I mean, am I some kind of cheap barroom hustler? Rather, it was my undeniably intense urge to correct misconceptions, my desire to clearly point out what is true and what is not, and my compulsion to educate and enlighten the populace. That, and my exasperation at the loudmouth jerk sitting at the next table.

I was at the Rathskeller, in deep consultation with the members of the Greater Miami New Wave Opera, an organization dedicated to the principles of convolution in relation to subhuman behaviorism. Their latest production, the metaphysical video "Studio Police," had broken new ground in the study of visual/verbal abuse with its vivid scenes of death and destruction symbolizing the impact of out-of-calibration tape machines on young minds. The members were also formulating a new theory of heuristic thinking as applied to the question of measuring the significance of auto-location against mutual coupling, to determine which was the most powerful tool. Clearly, they represented a startling new breed of social philosophers.

Meanwhile, our iconoclastic discussion was suffering from the effects of a blabbermouth at a nearby table. He was one of those know-it-all types—surrounded by admiring freshman girls—spouting this incredible nonsense on whatever technical topic came to mind. We listened with disgust as he launched into a discussion of digital recording—explaining all of these gee whiz things about digital—how all the Beatles' recordings could be stored in a space no larger than a period at the end of a sentence, how the cost would be so low that every man, woman, and child could own a complete library of every piece of recorded music, etc. The GMNWO members were shaking their heads. They hold a healthy respect, and contempt, for current digital methods; they realize that digital is far from perfect, they understand that with digital recording and reproduction we have merely begun a new phase of evolution. Then the Mr. Know-It-All launched into a fantasy description of

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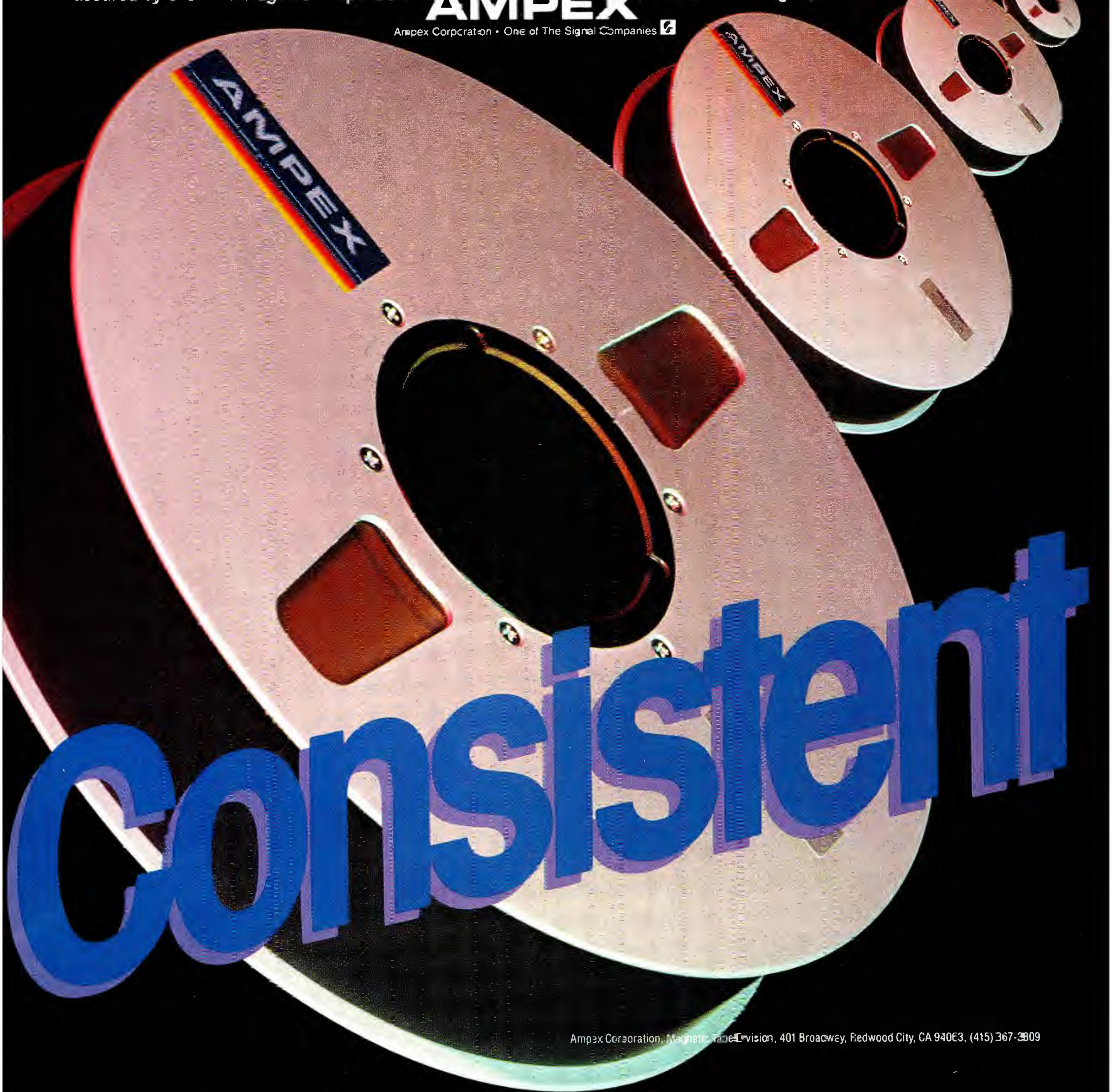

Ampex Corporation, Magnetic Tape Division, 401 Broadway, Redwood City, CA 94063, (415) 367-3809

how floppy disk digital recordings are comprised of 1s and 0s. I couldn't take it—something inside of me snapped. I pulled a floppy disk from my inside pocket and flicked it over to his table. He stopped in mid-sentence.

THE UNVARNISHED TRUTH

"Listen, you, that's a floppy disk—digital data—and there are no 1s and 0s on there, none at all. There never were and there never will be. Anyone who says there's 1s and 0s on there is full of dither."

"What are you talking about? I know what's digital! And every digital recording has bits in it, and every digital tape is just a lot of 1s and 0s. If you don't know that, you better get your batteries recharged."

He made a stupid face and all the freshman girls laughed. It was tragic to see someone so eagerly hang a "kick me" sign across their backside. One of the GMNWO members bloodthirstily thrust a piece of chalk into my hand. That supreme symbol of exposition pushed me over the edge.

"No, you're wrong about that. The technique used for digital recordings is called saturation recording, and it is considerably different from the way you have been representing it. The format of the data recorded on the medium is a function of the host system and varies from system to system, but the essence of saturation recording is current flow, not voltage levels as you imply. Let me show you how it works—we'll take this floppy disk as an example. The data is recorded in terms of bit cells in which the data is interleaved with clock pulses. By definition, each bit cell is the period between leading edges of clock pulses. Here's what a bit cell looks like." (See FIGURE 1.)

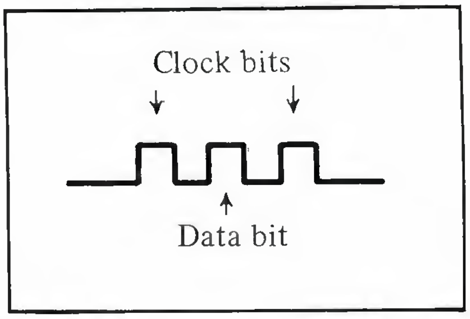

Figure 1. A bit cell, defined as the period between leading edges of clock pulses.

"That's just what I said! See—it's 1s and 0s."

"Control yourself. This is what the computer interprets, but the actual recording is something quite different. Can I continue? Each data bit has its own clock bit. Thus we refer to this type of recording as a frequency modulation (FM) recording. Data is written and read from the disk in serial form, and eight consecutive bit cells comprise a byte. The first cell is numbered 0, and the last is numbered 7. Therefore, when reading or writing, the most significant bit is transferred first. Look, here's a part of a bit stream, representing 10001001." (See FIGURE 2.)

"It still looks like 1s and 0s to me."

"Don't be naive. The standard FM encoding format can be altered to obtain modified, modified frequency modulation (M^2FM), which results in a doubling of the data transfer rate, to around 500 KBS. Error rate performance can be maintained through the use of phase locked loop data separators and write precompensation; most systems already incorporate these provisions. I'll explain that later, but first look at an M^2FM data word, the same 10001001 as above (FIGURE 3). For FM, the data bits are always written in the cell center, and a clock bit always leads the cell. For M^2FM, data bits are still written at the cell center, but there isn't always a clock bit. Specifically, a clock bit is written only if there isn't either a data bit or clock bit written in the previous cell or there is no data bit written in the

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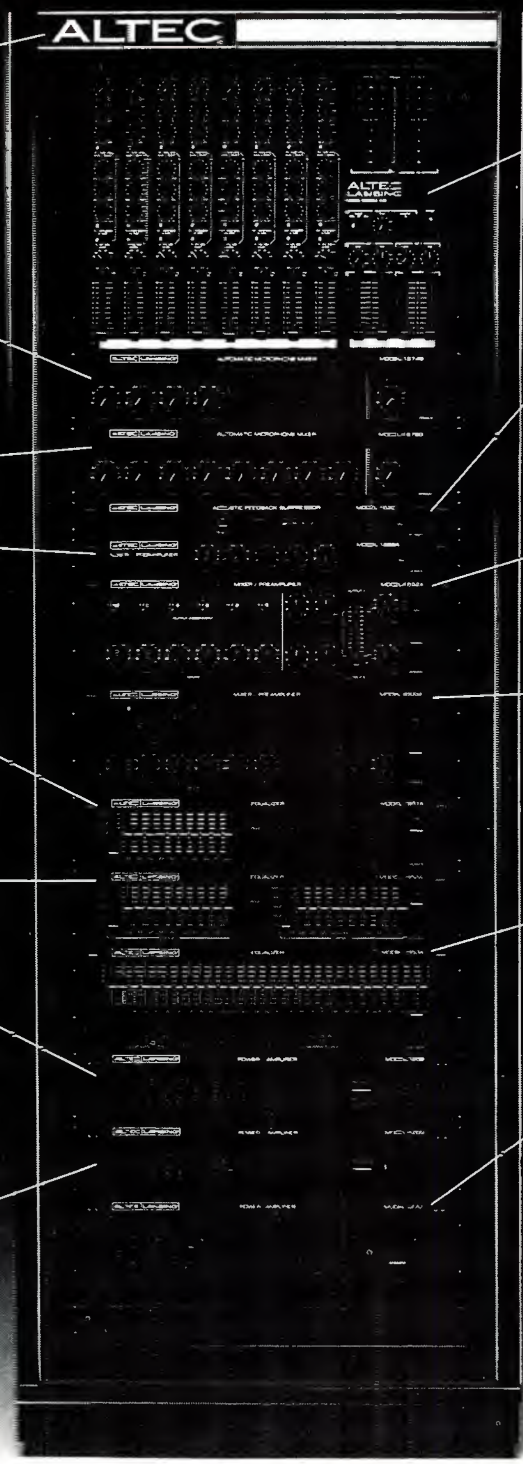

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Circle 23 on Reader Service Card

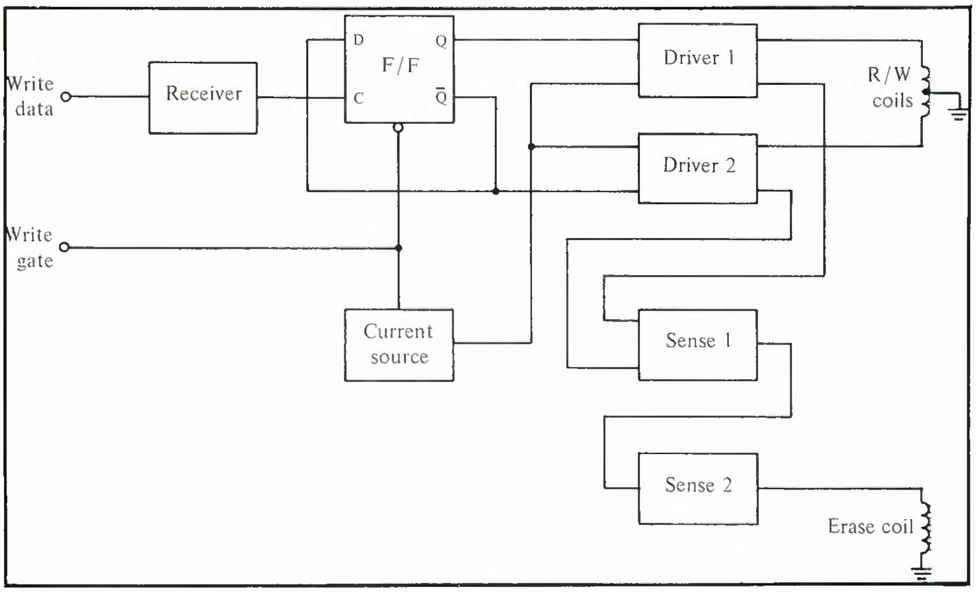

Figure 5. Write circuitry.

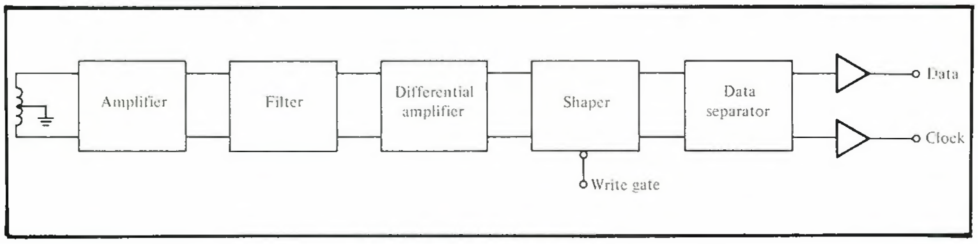

Figure 6. Read circuitry.

thus changing the direction of current flow through the head. As I mentioned before, this also erases old data. Look at this drawing (FIGURE 5). Write data cells are supplied by the host system and cause the write data trigger flip flop to toggle between the two write drivers. Thus current is alternately applied to each half of the read/write head. The write gate, from the host, clocks the flip flop and subsequently supplies write current; when write current is sensed flowing to the write drivers, a signal is generated to provide erase coil current.

"In the read mode, current is induced into the gap and one of the coils of the read/write head as the disk passes underneath, thus generating a voltage. When the next bit passes by, a current is induced in the other coil, causing another voltage pulse. The read circuitry looks like this (FIGURE 6). When the head is loaded, and the write gate is inactive, the read signal is amplified, de-spiked, and fed to a differential amplifier. With FM, a clock pulse occurs every four microseconds, and once every two microseconds when logical one data is present. Thus, the frequency of the read data varies. Specifically, the read signal decreases as the frequency increases. The shaper squares the sine waves from the head into equal amplitude pulses. The data separator, referencing two time constants, creates two data windows to distinguish between clock and data pulses. When the previous bit cell had a data pulse, a short window of 2.9 microseconds is used; the long window of 3:1 microseconds is used when the previous cell had no data. The clock window takes up the remainder of the cell time, 1.1 or 0.9 microseconds. The separated clock and data pulses are fed to drivers, and back to the host computer.

"Finally, the essence of the recording method can be explained. Shugart and other floppy manufacturers use the non return to zero (NRZI) ASCII recording technique. Its simplicity is beautiful. The read/write head is a ring with a gap and a coil; in the write mode, current flows through the coil, flux induced in the ring fringes at the gap, and this flux magnetizes the magnetic surface passing underneath. During a write operation, a bit is recorded when the flux direction is reversed by reversing the current in the coil. Hence the flux on the disk magnetic surface instantaneously changes polarity. It is this flux reversal that represents a bit on the disk.

"When reading, the disk flux reversal reverses the ring flux, and therefore the current induced into the coil. The gap first passes over an area that is magnetized in one direction, and a constant flux flows through the ring and coil. When a recorded bit passes underneath, the flux in the ring and coil makes a 180 degree reversal. It is this flux reversal in the coil which ultimately causes the voltage output pulse. Look—it's obvious when you look at a head, and its timing diagram (FIGURE 7). You see, there's no carrier, no bias signal, just the signal itself at any given instant in one of two states—full positive or full negative. The current flow through the head alternates direction, and we pass enough current through the head to magnetize the medium as much as it can be; the magnitude of the recorded signal is that corresponding to saturation of the medium. The data is stored in terms of zones of constant magnitude, and it's the reversals between zones which actually embody the data. It's totally obvious—but not at all like you were describing it."

A PYRRHIC VICTORY

I put down my chalk, still hot from my professorial high, and looked up from the table's scribbled diagrams. The members of the GMNWO were slumped over the table, blitzed, surrounded by empty dark Becks' beer bottles. The girls were gone. The know-it-all had moved on to another table; over the hubbub I could hear him explaining to somebody how EPROMs work—his description was totally off the wall. But I was vindicated; I felt secure in the knowledge that I had stated the facts. It isn't 1s and 0s. It's + and –. That's a big difference. ■

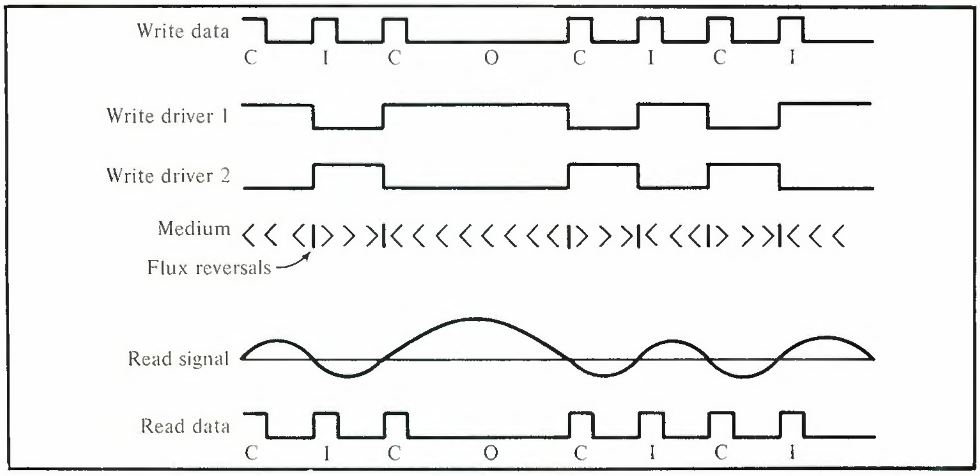

Figure 7. Read/Write data and timing diagrams showing the effects of flux reversal.

JOHN EARGLE

Sound Reinforcement

Microphones in Sound Reinforcement

Ed. Note: Due to an editorial lapse (read a few too many three martini lunches), Part III of John Eargle's column on microphones ran before Parts I and II. Therefore, disregard everything you read in last month's column until December. At that point, take your September issue out of your **db** *binder (you do have one, don't you?) and voila, you're in sequence. Sounds simple, huh?*

• Thus far in our coverage of sound reinforcement, we have concentrated on loudspeakers and environmental conditions. In this and the next couple of columns we will deal with that very important first element in the electro-acoustical system, the microphone. This month, our discussion will cover microphone specifications, including sensitivity, noise floor, dynamic range and operating impedances.

SENSITIVITY RATINGS

Unfortunately, there are quite a few ways of indicating the sensitivity of a microphone—many of them going back to a much earlier era in our industry.

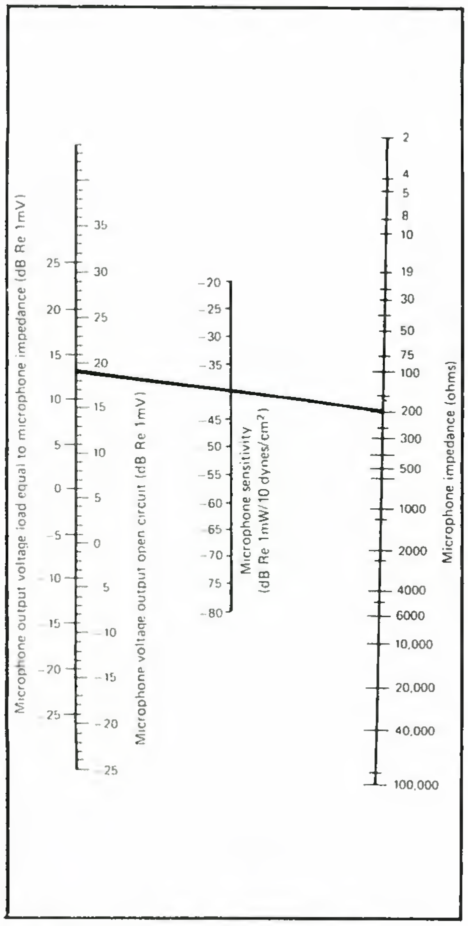

Figure 1. Nomograph of microphone impedance, output power, and voltage level.

Perhaps the simplest method is that which has been adopted by many of the European manufacturers.

Taking an example from a Neumann specification sheet, the sensitivity of the KM 83i capacitor microphone is given as:

$$7 \text{ mV/Pa} \pm 1 \text{ dB}.$$

What this means is that when the microphone is placed in a sound field of one Pascal, which is 94 dB SPL, the open-circuit output voltage is 0.007 volts, ±1 dB. Since most microphones operate essentially unloaded, this value will be quite accurate in any subsequent calculations of system gains and losses which might be made.

Now, let us look at the sensitivity rating of the Electro-Voice 635A dynamic microphone. It is given as

$$-55 \text{ dB } (0 \text{ dB} = 1 \text{ mw}/10 \text{ dynes/cm}^2).$$

Since this specification gives us the power output of the microphone in a given sound field, we must relate that power to a particular impedance if we are to find the actual voltage output. The published impedance is 150 ohms.

First, let us look at the sound field the microphone is placed in. Fortunately, 10 dynes/cm^2 is the same as one Pascal. (Note that the Europeans have converted to SI units, while EV, and many other US companies, still make use of CGS units.)

The power (P) corresponding to -55 dB can be solved for from the following equation:

$$10 \log P/.001 = -55 \text{ dB}.$$

Rearranging and solving for P, we get:

$$P = 3.16 \times 10^{-9} \text{ watts}.$$

This is a very small power indeed, and now we wish to find the voltage across 150 ohms which gives us this power:

$$E = \sqrt{(3.16 \times 10^{-9} \times 150)},$$

Solving:

$$E = 0.6 \text{ mV}.$$

This is the voltage which exists when the microphone is loaded with 150 ohms and placed in a sound field of one Pascal.

Now, let us unload the microphone so that we can compare it directly with the KM 83i. When we remove the load, the output voltage will increase 6 dB in level; that is, it will double to a value of 1.2 mV. It is not as sensitive as the capacitor microphone, since it is a passive device. On a voltage basis, it is some 15 dB lower in output than the capacitor microphone.

The nomograph shown in FIGURE 1 will enable the user to make the above calculation by inspection. Note, however, that the output voltages, loaded as well as unloaded, are given in dB levels relative to one millivolt.

In using sensitivity specifications, the designer takes into account the maximum levels that will exist at the microphone and calculates the voltages which will correspond to these. The overload voltage to the input stage is noted, and care must be taken that the voltage from the microphone will not exceed it. If there appears to be a problem here, then some kind of input padding must be used to cut down on the microphone's output voltage. Many capacitor microphones have switchable pads, usually 10 or 15 dB, that effectively reduce their voltage to the same range as that of a dynamic microphone, and the problem is usually avoided.

We will not discuss here the rather archaic EIA method of rating microphone sensitivities, since it is not encountered often these days.

MICROPHONE NOISE RATINGS

Normally, the effective noise level of a dynamic microphone will not be stated. On occasion, you will see a rating which states the hum susceptibility of a dynamic microphone when it is placed in a reference magnetic field. Such ratings are rare, since good shielding is almost taken for granted these days.

All professional-grade capacitor microphones will have a noise rating. Most of these state the equivalent level that would be generated if the microphone were placed in a stated A-weighted sound field. For example, a noise rating of 17 dB(A) indicates that the microphone has an electrical noise floor equal in level to that of a perfect (noiseless) microphone placed in a sound noise field of 17 dB as measured on the A-scale of a precision sound level meter. FIGURE 2 shows the A-weighting curve used for these measurements.

A knowledge of the noise floor of a microphone will determine its applicability in certain kinds of work. For example, the recording of fairly quiet instruments in very quiet environments at some distance may tax the noise performance of many otherwise good capacitor microphones. Typically, those microphones with smaller diaphragms will have higher noise floors than those with larger diaphragms. As a rule, we will not encounter these problems in sound reinforcement.

DYNAMIC RANGE OF MICROPHONES

The dynamic range of a microphone is set by the noise floor at one end and by some arbitrary distortion rating at the other end. A distortion rating of 0.5 percent Total Harmonic Distortion is usually taken as the upper limit. Thus, if a capacitor microphone has a noise floor of 17 dB(A) and reaches a distortion of 0.5 percent at a level of 120 dB SPL, then its overall dynamic range will simply be the difference between these ratings, or 103 dB. Thus, a good microphone will have a dynamic range greater than even the best digital recording systems in use today.

Since they are active devices, capacitor microphones will overload rather abruptly when the maximum rated level is reached. Dynamic microphones, on the other hand, do not usually carry a maximum sound pressure rating, since their linearity extends well beyond the normal sound levels they are expected to be used in. Certainly for most applications in sound reinforcement, they will be picking up sounds that lie well within their useful dynamic range.

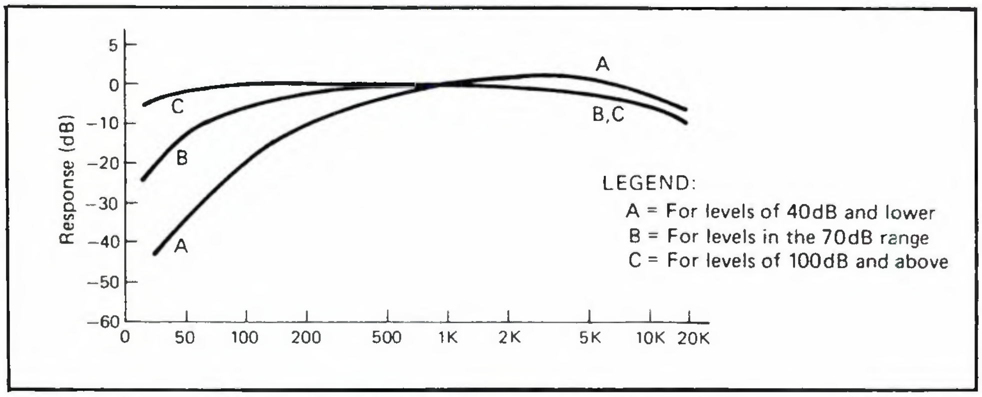

Figure 2. Weighting curves for sound level measurements of (A) 40 dB and lower, (B) in the 70 dB range, and (C) 100 dB and higher.

IMPEDANCES

Nobody interested in professional sound reinforcement work should bother with high-impedance microphones. The term *low impedance* usually implies that the microphone has a source impedance anywhere from 50 to 200 ohms. In most mixer inputs encountered today, these microphones will see an input impedance usually about ten times that of their source impedance, and thus we can say that they are effectively operated "open circuit."

Most capacitor microphones will state the minimum impedance into which the microphone should work. Dynamic microphones can of course be operated into a matching load. ■

LEN FELDMAN

Sound With Images

Stereo and High Definition TV Update

• It's been almost a year since I reported to you on the status of stereo TV broadcasting in the United States. I stress the word *broadcasting* since, for some time now, cable operators have been supplying high-quality stereo sound to their subscribers using a variety of systems, most of them involving the use of an FM stereo receiver or tuner as an adjunct to the TV set. Cable operators, after first receiving the audio signals from satellite downlinks, translate the left and right recovered audio signals to a composite stereo multiplex signal similar to that used in stereo FM broadcasting. They then use that signal to frequency-modulate a carrier that is piped down their cable to subscribers. The frequency selected for this carrier is one which falls in the regular FM frequency band, between 88 and 108 MHz. In short, from the subscriber's point of view, TV stereo today is no different from simulcast broadcasts which are initiated by some TV stations in partnership with existing FM stereo stations, where both a TV receiver and an FM stereo tuner or receiver are required to receive the programs in stereo sound.

It is now nearly five years since Japan began broadcasting TV stereo using a system developed by NHK, the Japanese broadcasting authority. That system uses (L + R) and (L – R) matrixing, in which the (L + R) (mono equivalent) is transmitted on the regular main audio RF carrier, using FM modulation, while the difference information (L – R) is used to *FM modulate* a sub-carrier which is set at twice the horizontal line frequency, or at around 31.5 kHz. In this country, that same system has been proposed by the EIAJ (Electronic Industries Association of Japan). A Multi-Channel Sound Committee, under the administration of the Electronic Industries Association (EIA), has been working for several years to develop a technical record concerning the performance of this basic system as well as of two other transmission systems: one by Zenith Radio Corporation, the other by a small Chicago firm called Telesonics, Inc. Both of these latter systems employ an *AM Sub-carrier* rather than the frequency-modulated sub-carrier proposed by EIAJ and used in Japan. Furthermore, all three U.S. proposals make provision for a so-called S.A.P. (secondary audio program) channel, which might take

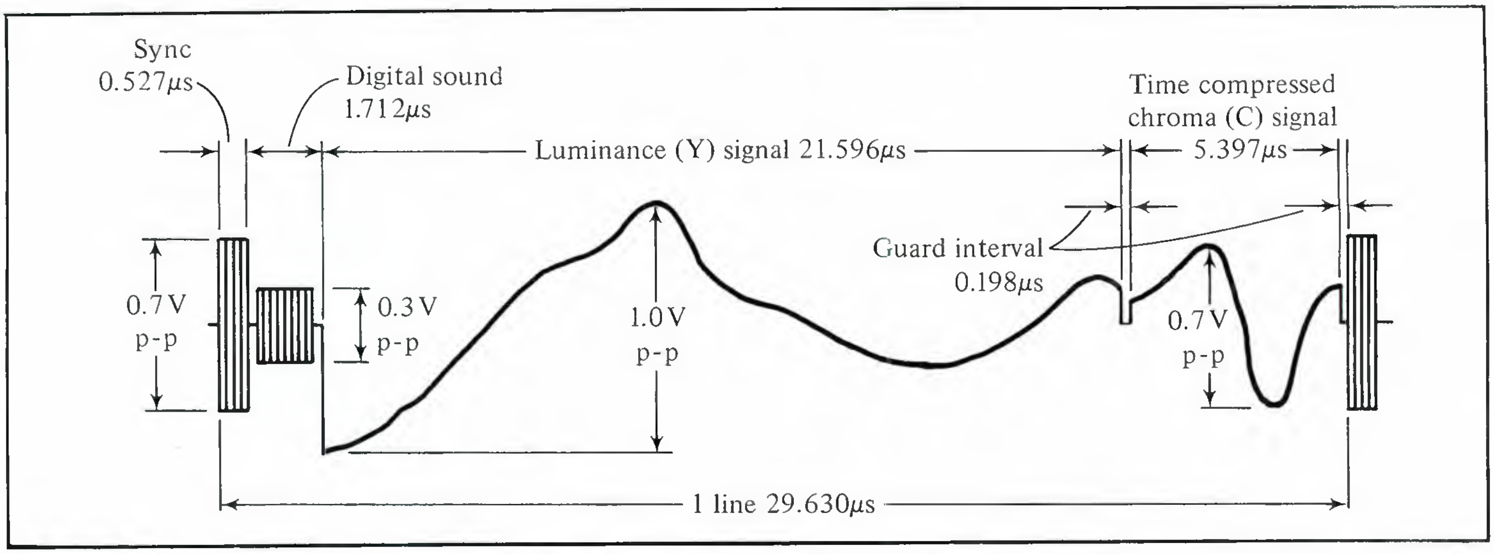

Figure 1. The method NTT and NHK propose to compress high-definition video and digital audio signals into a limited bandwidth channel of 15 to 20 MHz.

the form of a second-language transmission in metropolitan areas that are heavily populated by non-English speaking minorities, or might be used for totally unrelated audio material.

Further complicating the selection process is the established need for some form of companding to compensate for the degradation in signal-to-noise (for all three transmission systems) when stereo audio is transmitted. In the case of FM stereo, this loss of S/N can amount to as much as 23 dB under weak-signal conditions. The situation is not quite as bad in TV audio; an average loss of S/N amounts to 15 dB when the switch is made from mono to stereo. However, it must be remembered that TV audio has a 10 dB disadvantage to begin with, since maximum frequency deviation of the main audio channel in the NTSC system of TV transmission is 25 kHz, as opposed to the maximum modulation of 75 kHz in the case of FM stereo.

There are now five separate proposals for companding (noise reduction) systems to compensate for this loss of signal-to-noise ratio. These, too, must be thoroughly tested by the committee, who are going about their task by conducting a series of listening tests using experienced auditors.

PRESENT TIMETABLE FOR TV STEREO

If all goes according to plan, a final report should be available for submission to the FCC by late fall of 1983—perhaps even by the time you read this. The hope is that this report will recommend a single, specific system—that is, a combination of one of the three proposed transmission systems and one of the five proposed companding systems. This will be a departure from previous EIA Committee practices which, in the past, have simply presented comprehensive data to the FCC and waited for that regulatory body to choose "the best system."

In recent years, the FCC has done a complete about-face and is now in a mood of deregulation. This was amply demonstrated when the FCC refused to choose one of five proposed AM stereo systems, preferring instead to "leave it up to the marketplace." This so-called "marketplace" decision has resulted in a failure of AM stereo to get off the ground, much to the disappointment of AM broadcasters and receiver manufacturers. To forestall such an event from occurring in the case of TV stereo, the Committee hopes that by sending a specific recommendation to the FCC, the system chosen will become a de facto standard even if the FCC goes "marketplace" again—as they are expected to do.

The timetable is very critical, however, from a legal point of view. Once the FCC opens a Docket in the matter (expected, at the latest, in September 1983), there may be a response period of as little as 90 days. Shortly after that, the FCC might well issue its report-and-order, setting broad parameters for TV stereo audio and, again, leaving the choice of a final system up to the marketplace and the competitive free-enterprise system. As of that moment, it would become illegal for the various proponents to sit together in committee as they are now doing; that would constitute a violation of the anti-trust laws of the United States. It's for this reason that the EIA MTS Committee is pushing hard to complete its work before the end of 1983; and that need to complete the multi-channel TV report in short order could hasten the day when TV stereo broadcasting is finally initiated in this country.

HIGH DEFINITION TV IS MAKING PROGRESS, TOO

While we get ready to improve the sound quality of TV broadcasting, Japan is off on another project aimed at improving the picture quality of TV. For at least three years now, several Japanese companies have successfully demonstrated so-called high-definition TV, which generally involves pictures having 1125 lines and an aspect ratio that's closer to that used in wide-screen motion pictures. The pictures are literally as sharp as those projected from 35 millimeter film, which means that they particularly lend themselves to large screen projection TV displays. Such large screen TV, of course, would fit in very nicely with the coming of stereo audio for video, since there are still many who feel that wide separation in audio for TV, coupled with relatively small-screen viewing, may prove disconcerting to many viewers.

While closed-circuit demonstrations of high-definition TV have been extremely effective and impressive, the problem remained as to how to transmit such video information from the TV studio to the viewer's home receiver. Once again, NHK of Japan seems to be at the forefront of research—this time directly aimed at solving the bandwidth problem involved in HDTV. The network had originally hoped to get on the air using transmission channels in the 12 GHz band, but so far such allocations have not been granted to it. Had NHK been able to transmit in that frequency range, the full 30 MHz bandwidth (per channel) needed for 1125 line TV would have been no problem. As matters stand, NHK is working on various approaches to squeezing down the

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bandwidth required. In one approach, a "frame memory" at the receiving end would convert conventional moderately-high-definition interlaced signals into line-sequential signals. Instead of using every other 1/60th of a second field to show alternate lines of the 1125 lines that make up a high definition picture, the proposed receiver would display fewer lines—perhaps 875—but would display all of them sequentially in every field scanned. It has been shown that the perceived resolution of a line-sequential video display is about one and a half times better than that of an interlaced display, given the same total number of lines.

At the transmitting end of the system, NHK would convert 1125 line high-definition video signals into the required 875 lines by using equipment similar to that employed to convert European (PAL System) 625-line pictures into 525-line pictures now seen in the U.S. and Japan. The luminance signal (brightness) bandwidth for an 875-line picture works out to be around 15 MHz. By compressing the chroma-signal by a factor of four-to-one, it is possible to maintain a total bandwidth for both the luminance and color signals of no more than 15 MHz, or half the bandwidth that would have been required for a full 1125 line high-definition TV picture. This sort of compression, called time-compression multiplexing, can be done because the luminance signal occupies only about 73 percent of the time allocated to each transmitted line. Normally, the chroma signal could not fit into the remaining time on each line, because it occupies and fills the same time slot as the brightness or luminance signal. The chroma signal's maximum *frequency*, however, is only 5 MHz, so if its *time axis* is compressed by a factor of four-to-one, it can be appended to the end of each scanned line.

To accomplish this type of bandwidth compression, the chroma signal is first converted into a digital signal and stored in a digital memory. Then it is read out of memory at a clock rate that is four times as fast as that used for sampling the analog signal and finally converted back to the compressed analog format.

FIBER OPTICS FOR HDTV TRANSMISSION

Another quasi-governmental agency in Japan, the Nippon Telegraph and Telephone Public Corp. (NTT), has found a practical way to transmit high-definition video signals through optical fibers directly to the homes of future telephone subscribers. In one approach to such a system, all channels in a new type of "CATV system" would be switched at the local telephone exchange. While trunk transmission would be digital on the optical fiber that would connect subscribers to exchanges, transmission would be analog. In this system, NTT engineers would squeeze the normally required 30 MHz bandwidth (luminance, chroma, sound and sync) into the 20 MHz bandwidth required by the luminance signal alone, using the time-compression multiplexing technique described earlier. In the case of the NTT system, ample time remains on each line for a time-compressed PCM (pulse-code modulated) digital audio signal with a clock rate of about 15 MHz, as well as for a sine-wave sync signal. FIGURE 1 illustrates the various signal components of a single line of compressed video and audio information.

Although the U.S. led the way in TV broadcasting, it is clear that other industrialized nations of the world are fast catching up with us and even surpassing us in the field of entertainment electronics. I can clearly remember when my good friend William Halstead, a noted consulting engineer, was called upon by the Japanese government way back in the 1950s to help them get started in TV broadcasting. For better or worse, Mr. Halstead steered NHK into the NTSC system, the same system that we are straddled with in the U.S.; Europe opted for the higher resolution PAL system. As far as I know, NHK never complained to Mr. Halstead or questioned the wisdom of his choice. Perhaps this advanced work in high-definition TV is their "inscrutable way" of getting back at us for talking them into the NTSC system in the first place. ■

BARRY BLESSER

Digital Audio

Applications of Re-sampling

• During the last two months, we have presented the basis for building a digital re-sampler. This is a device that converts a sequence of digital audio data samples from one frequency to another. We demonstrated that the basis for this approach is that of creating an internal sampling frequency which is the least common integer denominator of the two frequencies. For example, to convert from 50 kHz to 40 kHz requires us to find a frequency such that the 5:4 are integers. This would be 200 kHz. As the ratios become more irrational, the internal frequency becomes higher and higher. Thus, the conversion from 44.1 to 48 kHz results in a ratio of 147:160. The effective internal frequency is thus 7.056 MHz. This is rather high for most hardware.

A second more interesting issue is that of clock synchronization. The rate at which numbers pop out of the re-sampler is fixed based on the internal divider ratios. However, the absolute time is not fixed. In the above example, there will be 160 input numbers at 44.1 kHz input rate for every 147 numbers at 48 kHz. This rate is unambiguous, but there is no implicit mechanism for creating the output clock. Consider the issue in the following terms. When the 1st (nominal) input sample enters at 44.1 kHz, we define the 1st (nominal) output sample at 48 kHz. The 160th input sample will enter on the 160th input clock cycle and the 147th output sample will exit at that time. The 2nd input sample will enter on the 2nd clock edge of the input clock which will be 22.6757 μsec after the first input sample. We expect the second output sample to appear 20.8333 μsec after the 1st input sample, corresponding to the expected 48 kHz clock period. However, we have not indicated where the 48 kHz clock information comes from. The 44.1 kHz input clock is assumed to come with the 44.1 kHz input data.

We would appear to have two choices for the 48 kHz output clock. On one hand we can assume that the re-sampler electronics could also create such a clock to appear with the output data; on the other hand we can assume that there is some master system clock at 48 kHz located in the studio. Each of these assumptions has its problems.

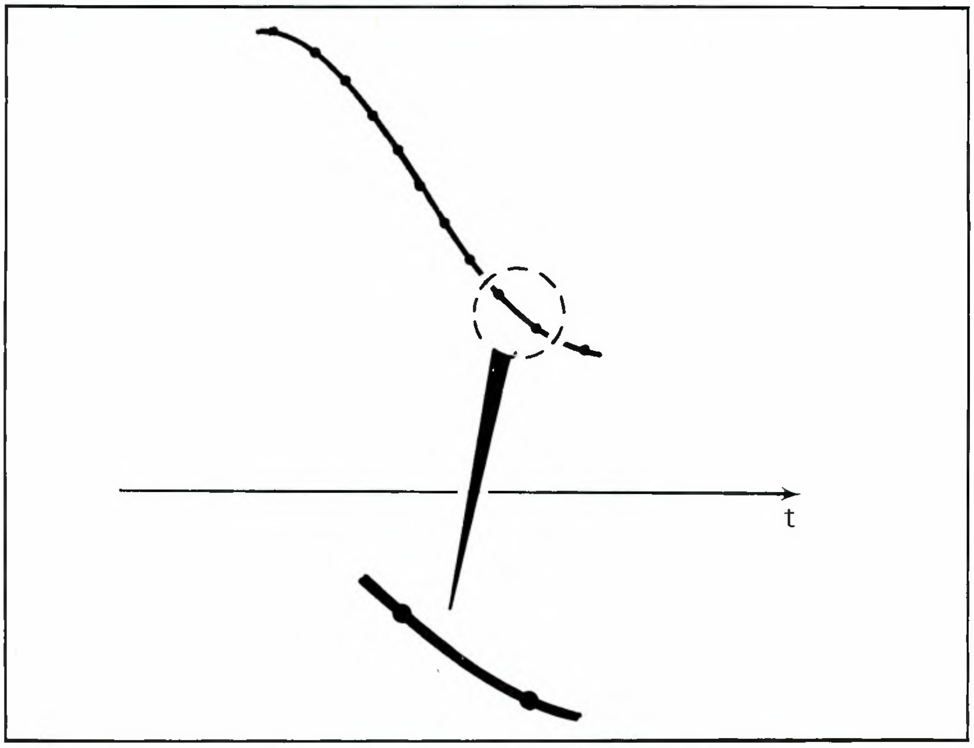


Figure 1. Segment (top) of 20 kHz audio after re-sampling at a very high internal sampling rate. Two samples (below) of the signal showing that linear interpolation is very accurate because of the low value of the higher order derivatives.

CLOCK GENERATION

We take the case of clock generation first. Who can create a 48 kHz clock out of a 44.1 kHz input clock? In a real sense this is the frequency synthesis issue, and we can refer to the techniques used for this kind of process. One such approach is to build a series of frequency multipliers that will convert the 44.1 input clock to a 7.056 MHz. Since the factor is 160, we could build a set of multipliers using factors of 4, 4, 5, and 2. A frequency multiplier is nothing more than an impulse generator running at the input frequency followed by a filter that filters for the nth harmonic. For example, if the input clock triggers a 50 nsec monostable, we will have a periodic impulse train at 44.1 kHz. A filter placed at 220.5 kHz will filter the 5th harmonic. After suitable filtering, the result will be a sinewave which can then be used to feed the next frequency multiplier. In the end we will have created the 7.056 MHz sinewave which can then be divided digitally by a factor of 147.

Although not a trivial activity, we can see a mechanism for doing the task. The problem gets more difficult as the ratios become more irrational. Another technique for creating the clock generation is that of using a phase locked loop. This is an advanced signal processing idea which is beyond the scope of this article and may be discussed in the future. Regardless of which method is used, we should note that the allowed phase jitter on the clock must be very small if we are to preserve the required audio signal quality. Jitter on the order of 1 nsec will degrade high frequency audio energy way beyond the 98 dB of a pure 16 bit system. The technical difficulties with the jitter problem often rule out the use of simple clock synthesizers. In fact, phase jitter is a major intellectual problem in all frequency synthesizers. There is extensive literature which shows that the task is very difficult and requires a very high level

of sophistication. The average audio hacker would not want the problem.

Now that we have gotten to the point where we understand that such a converter could be built, we are left with an interesting system issue. The clock regeneration now becomes the master for all activities which relate to the output data. Assume that there was other equipment in the studio running from the studio master clock system. Its frequency is also specified at 48 kHz. The 48 kHz clock regeneration must be exactly the same as the studio 48 kHz. The two can only be made exactly the same if there is a wire between them: i.e. one is master, the other slave.

If they were independent, we would not expect them to be exactly the same. Suppose one was 48.0001 kHz and the other was 48.0000 kHz. That means that there would be 48,001 samples of one clock for every 48,000 samples of the other. We can no longer connect the audio data across this boundary because we will need to repeat one sample twice or to ignore one sample every 10 seconds. This is not a large degradation but it is one which we would like not to build into the basic structure of converting between sampling frequencies. The slippage between the systems is not ideal.

We are now stuck in an unsolveable problem. On one hand we wish to use the ultra-high quality studio clock reference because it is designed for low jitter. On the other hand, we need to make the output clock of the re-sample be an exact integer ratio to the input frequency and an exact match to the studio clock.

One of the papers at the Rye Digital convention presented by Roger Lagadec proposed a very novel approach to this problem by allowing the re-sampler to have two clocks: an input clock and an output clock. At first look this appears to be relatively simple. With a second look, we see that this is not so simple because the re-sampling is no longer done at a predetermined rate. If the system can respond to small changes in one of the clocks (desired characteristic), it could also respond to larger changes. In other words, the system provides an output sample on demand.

The exact details of this approach have not been presented publicly because of patent issues. Until such time as the patents are issued or rejected, we will not have the full story. The paper does provide some clues, however. The key revolves around the original idea of linear interpolation between samples, as discussed two months ago in this column. We rejected it at that time because linear interpolation is not very accurate except for low frequencies. At low frequencies, the errors are very small.

Thus the key to the approach is to create an internal effective sampling frequency which is relatively high. At this high frequency, all audio information is relatively low frequency. Compared to 5 MHz, 20 kHz is very low. FIGURE 1 shows a section of a 20 kHz audio signal sampled at a very high frequency. The blow-up below shows the part of the signal between two samples. The segment connecting the two samples is almost perfectly linear. In fact, the higher the sampling frequency, the more linear the approximation in a square sense. Doubling the ratio lowers the error by 4. We can thus tolerate the error.

The final interpolator can thus be done at very high speeds and may in fact be performed analog. This is only a guess and we will have to wait for the results.

The interesting aspect of this solution is that it allows two digital systems to be interconnected without regard to the clock frequencies, phases or jitters. The result of re-sampling will be as good as the two clocks which the user provides. The need for the re-sampler in this application only comes from the fact that the industry has not standardized on a particular single frequency. There was a strong move to create a standard from the beginning, but there were very conflicting constraints.

SAMPLING STANDARDS

The arguments for sampling frequencies were extensive. One group wished a low frequency to achieve higher channel usage. The European broadcast community wished a low rate in order to get more audio channels on their digital multiplexed carrier systems. They also wished simpler and cheaper hardware that comes from lower rates. Led by the BBC and German Post Office, a series of experiments were conducted on the required audio bandwidth in order to determine the sampling frequency. The results clearly showed that 15 kHz of bandwidth was more than adequate in any real listening situation. The results are available in technical journals and in the BBC engineering reports. With this bandwidth, they selected a sampling frequency of 32 kHz.

Another group led by the Japanese Consumer Electronics Industry wanted frequencies that were compatible with TV format. They believed that digital audio would share common hardware and perhaps use the TV format for digital bits. They also considered having digital audio with video. The result of conflicting demands led them to 44.1 kHz.

Finally, the professional audio group wanted a higher frequency and one that was compatible with other references including 50 and 60 Hz. The choices eventually reduced to 50 and 48 kHz with several other possible choices. The 48 kHz was especially interesting because of the 3:2 ratio with 32 kHz. However, once the standards issue became more clear, with the professional using 48, the broadcast group considered switching. We are thus left with a consumer standard and a professional standard.

During the last 12 months there was extreme pressure placed on the consumer group to switch to 48 kHz. In the end, they elected not to switch. The public reason given for this is that the development was too far along to switch. It is unclear if this is the real reason. We may find out that there were compelling political and economic justifications. For one thing, it makes it difficult for professional equipment to be used in the consumer environment without the special re-sampler. Perhaps this makes piracy more difficult.

Any tape recorder which can record at 44.1 kHz would, presumably, only be useful for "taking" copies of copyright digital audio discs. Even a broadcast station would require either a sampling rate converter or a special issue of the audio disc in order to process it using professional equipment. The converter proposed by Lagadec is not a simple piece of equipment, with, we presume, a price of $5 to $10 k. Rationalism is not always the driving force in a technical industry. One may draw the analogy to the early days of long playing records where there were more than 5 different pre-emphasis standards. A typical preamplifier had a knob to select which one to use. The analogy breaks down, however, because the cost of different pre-emphasis frequencies is reduced to a pair of R-C time constants.

With digital audio, the costs are extremely high. One may hope that a long term evolution of the audio disc could result in a convergence into a single sampling frequency standard. At this time, the only possibility comes from a little known fact. The production of digital audio discs may be limited by the choice of formats. The bit size is sufficiently close to the wavelength of light that it may be very difficult to manufacture economically. This might open the door to a second standard at a later time—with a large diameter and a large bit size. Such a second standard might come when pressure in the professional end of the business results in music in the 48 kHz format.

By the time this article reaches you the reader, the situation may be much clearer. At the time of writing, I can only speculate that even the best laid plans of engineers and businessmen do not always lead to the obvious conclusion. For those of you who remain somewhat cynical, I suggest that you adopt the slogan: "Remember Quad!" ■

Conventions and Conferences—More Than Semantics

LAST MONTH after an absence of some fifteen years, the Audio Engineering Society returned to the New York Hilton to convene its 74th convention. During this interval, the Society's East Coast conventions have been held mostly at the Waldorf-Astoria, which is now a little too small to contain the show. In fact, even the Hilton proved to be a tight fit, and at least a few last-minute exhibitors were not able to be accommodated.

While this might suggest there's a market for more and/or bigger pro' shows in the future, it seems the next U.S. gathering may actually be quite a bit smaller. At the last European convention, the Society's governors voted to return to the traditional two U.S. shows a year—one each on the West and East coasts. The decision was greeted with instant hostility by many exhibitors, who were suffering from acute convention burnout. Committees were convened, letters were launched, and the phones started ringing.

After some deliberation, the Society decided to stand by its two-show decision—almost. According to the latest word, the next U.S. gathering has been reclassified, from "convention" to "conference." Exhibit space will be limited, and the emphasis will be on the technical papers instead of the hardware.

Perhaps it's about time. According to the AES's charter, its gatherings are supposed to be "Technical Meetings and Professional Exhibits." In fact, the last "Convention" was held in Paris, in 1977. Since then, the former phrase has been used on the Society's programs distributed at the conven...oops, at the technical meetings.

This little bit of semantic subtlety was often lost on those who attended these gatherings, as they were clearly hardware shows, despite what you read in the program. Yet, as hardware shows, they were in conflict with themselves. Exhibitors would spend fortunes to drag in the latest toys, sprinkle their booths with personnel, and then promise not to sell anything.

Booths and budgets got bigger, and the tail began wagging the dog, as the technical papers were all but ignored by legions of tire kickers who came to admire, but not to buy, the hardware. Something had to give, and as often as not it would be the temper of an exhibitor, caught in the middle of audio's own Catch-22; stay away and be noted by your absence, or attend in full regalia and give your accountant an ulcer.

There's no easy answer to this dilemma, as noted by the on-again/off-again convention scheduling of the last year or so. Presumably, the Society itself has no preconceived notions on all this, and is simply trying to respond to the wishes of its members—once it can get a clear reading on what's really desired.

Of course, asking the membership what it wants is a risky business, since the Society is made up of individuals, and not of companies. Most of these individuals enjoy seeing the exhibits, except perhaps for the comparative handful who gets stuck minding the booths. If it came to a vote, the members who are directly responsible for their company's financial health are heavily outnumbered by those who are simply attendees at a convention.

So maybe it is up to the Society itself to set the policy after all. (Well of course it is, but we mean doing so without a clear mandate from the membership.) And maybe the concept of one "convention" and one "conference" is the ideal solution. At the convention, exhibitors can try to outdo themselves with bigger and better displays. At the conference, the authors can do the same with their technical papers.

Who will win this kind of contest? Perhaps it will be us members. Wouldn't that be nice?

AND ON A PERSONAL NOTE...

Traditionally, Editorials and royal decrees are written in the form known to English majors as first-person plural. This conveys the impression (to third-graders, at least) that more than one person is out there pontificating. We (you see?) don't know how this helps the various monarchs out there—the only monarch we (there it is again) ever met was a butterfly, and it didn't have a thing to say.

As for editors, "we" is kind of nice (are kind of nice?), since it lets them off the hook later on. "We" can always claim that "they" wrote those offending words while "we" were off doing something terribly important elsewhere. (If only we had known..., etc.)

Well, it's time for we—I mean, me—to shift back into first-person singular for a sentence or two. Effective with next month's **db**, Publisher Larry Zide resumes his dual role as Editor/Publisher. He's been vacationing long enough as Publisher only, and it's time for him to go back to work. I'll be sticking around to help with some of the technical chores, but will be able to devote a little more time to some outside interests, like sleeping, and trying to make a fortune without working for it.

It's been a constant challenge trying to keep up with a field which we—sorry, I—had been told was relatively slow-moving. If pro' audio is slow, I'm glad I don't drive in the fast lane. And for the next little while, I think I'll pull off to the side, and re-read some of that stuff I passed over the last few years. If I find any errors, it was probably "their" fault. JMW

Wurlitzer's Model 1015 of 1946. Note theatre curtains—for five cents you got a song and a show. (Photo by John R. Davis; courtesy Judith's Jukes, San Francisco, CA.)

Rock-ola's Magic Glo-Model 1422 of 1946 is a good example of the wooden cabinet that produced a resonant tone. (Photo by John R. Davis; courtesy of Judith's Jukes.)

Wurlitzer's famous Peacock Model 850 of 1940 was the first to use a polarizer to break up the liquid in the bubble tubes into two colors. One of Paul Fuller's most eagerly sought designs. 10,000 were made but only 1000 or so are left. (Photo by John R. Davis; courtesy of Judith's Jukes.)

Wurlitzer's Model 950. This is one of Paul Fuller's rarest models. Due to a shortage of materials, only 3,400 were shipped in 1942. Today it is one of the most difficult to find. (Photo by John R. Davis.)

CAROLE B. DAVIS

The Music Goes Round and Round

Interest in the venerable jukeboxes of yesteryear is making a comeback. What next, a top 40 hit for Patti Page?

A 1934 ADVERTISEMENT from the Wurlitzer Company shows a workman wielding an ax. All around him lay the smashed remains of older model jukeboxes. In those days it was common practice to break up old boxes to force the sale of new ones; a new jukebox back then sold for about $239. Today it costs upwards of $6,000 to restore the survivors of not only the ax but of dust, dampness and neglect as well.

THE COMEBACK TRAIL

Jukeboxes are staging a comeback, and old ones, no matter what condition they're in, are eagerly sought by collectors and investors. Yesterday's cast-offs are today's treasures, and curiously these relics of another time are prized today more as art objects than for their original use when they were "America's favorite nickel's-worth of fun." Besides, as an investment requiring many nickels, the juke's illuminated bubble tubes, with its softly swirling red, blue and green glows is a lot more fun to look at then coveted gold Krugerrands hiding in the family vault.

For those lucky enough to find some of the old 78 rpm records to play on their restored boxes, Patti Page's "Tennessee Waltz," the Andrew Sisters singing "Don't Sit Under The Apple Tree," Elvis Presley or Der Bingle himself might float out over a wall speaker bearing Wurlitzer's famed horned and hatted *Johnny-One-Note* logo of 1946.

During the heyday of the jukeboxes (from the mid 1930s to 1948) Americans went wild pumping nickels into these music machines. This meant profits for both the phonograph and the record industries. But why this mania with a coin?

FROM IDOLS TO EDISON

The concept of exchanging a coin for goods and services has been around for a long time. Consider the ancient Eastern temple where the eyes of the idol could be made to drip tears when the worshipper placed an offering on some portion of the idol. The weight of the offering triggered a valve connected to a vessel deep beneath the temple. This vessel was filled with hot water. When the valve was tripped, steam was forced up through the bamboo tubes leading to the idol's eyes. What power! One could make the deity cry and even control the amount of tears by the simple act of controlling the amount of the offering.

The same psychology exists in any coin-operated amusement machine—that is, a feeling of control; a one-to-one relationship between the patron and the source. And as a source of entertainment, the jukebox was tops. By 1937, 225,000 of them could be found in restaurants, pool halls, candy stores, coffee shops and diners across the nation. By the mid 1950s there was one machine for every 330 Americans.

Carole B. Davis is a freelance writer working out of Mountain View, CA.

It's rather humbling to realize that the whole thing began in 1877 when a stunned and astounded Tom Edison rotated the crank, reset the stylus on his crude machine and heard his own voice repeat, "Mary had a little lamb"—the very words he had just recorded on a piece of tin foil.

A few years later a wax coated cylinder replaced Edison's tin foil. The two scientists who did this merely turned the name of Edison's invention around and offered the public the gramophone. As is often the case, competition stirred up action. "Why can't people leave my talking machine alone?", fumed Edison, who was forced to make improvements on his machine, the phonograph.

SOME EARLY ATTEMPTS

In 1886 the nation's innkeepers latched on to a good thing. They discovered the penny-in-the-slot mechanism worked

Figure 3. Music Designers' studio, featuring seven speakers for audio playback.

each speaker location. This was necessary due to the varying acoustics with changing crowds, seasons and weather conditions. For this purpose, four Klark-Teknik DN22 series octave band equalizers were chosen. Other major pieces of equipment used for Timespell included four dbx 150 Noise Reduction units and four QSC 1400 amplifiers (see FIGURE 4). For the walk-in music for the show, the Sony auto-reverse cassette deck with Dolby C™ was chosen. In trying to keep the cost of the entire show down, it was decided that a cassette deck would be acceptable for the "mood music" at the beginning. Except for the dbx and Klark-Teknik units (which have spare channels), there are spare units for the other pieces of equipment on the site.

The Audio Service Department at PRS custom built a dual shelf-rack to accommodate the Otari and associated equipment.

THE SPEAKERS

The system specifications for the speakers called for them to be wide bandwidth, high efficiency, relatively compact in size, and weatherproof; it also called for "animal and insect proof, and the ability to withstand large pieces of falling rocks."

"Working with Kenton Forsythe of Eastern Acoustic Works, Inc. of Framingham, Massachusetts, the FR-350R speaker was chosen. This is a high-powered compact three-way speaker system built of baltic birch with a horn-loaded 15-in. speaker, a 1-in. compression driver, two tweeters and a heavy duty passive crossover.

The eight stock speakers were covered with fiberglass and fitted with protective covers. The crossovers were sealed in boxes. The front of each speaker was covered with a very fine stainless steel mesh, and the entire speaker was painted a neutral grey to blend with the rocky environment.

"The speakers were a good choice," said Ouellette. "Knowing their capability allowed us to achieve the effect we wanted. We wrote into the script exactly where the music came from, planning that the audience would experience a different sound depending upon where they were standing."

The sound system was built and tested in the Audio Service Department at PRS, then dismantled and driven to Watkins Glen early in April 1983. Blake and Patton of PRS engineered the installation.

THE INSTALLATION

During the winter months prior to the installation, the construction crew was occupied with building the main control booth. The original plan for the location of the booth was a distance away from the Gorge with the projector being the only equipment in the control booth. However, after designing the booth for just the projector, and then realizing the necessity for heat sync space and additional space for the air conditioner, the control booth grew big enough to house all of the equipment; the dimmers, lasers, projectors, computers, and the audio system.

The electricians from Kirton Electric Co., New York, had run over seven miles of wiring for the speakers and lighting instruments. The longest unbroken stretch of speaker wire is approximately 350 to 400 feet. By using a 10 gauge wire

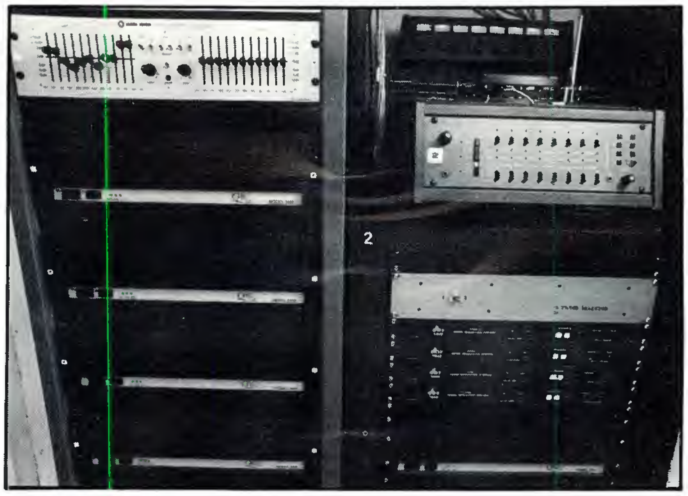


Figure 4. A portion of the audio rack designed by PRS Service Department.

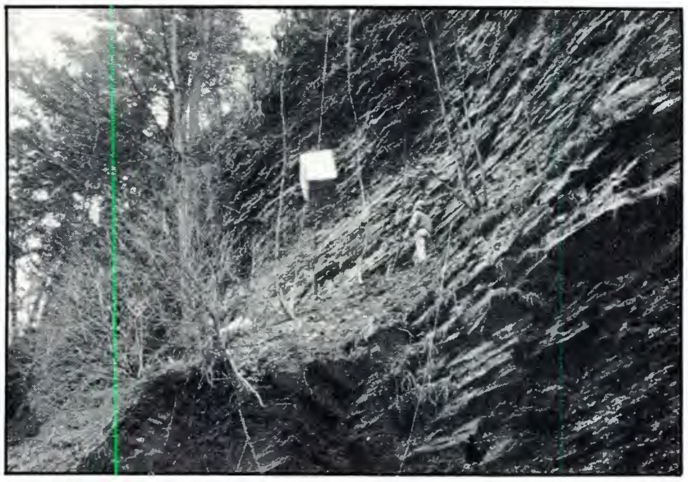


Figure 5. During the precarious installation, 160 lb. speakers were hoisted and positioned within the 110-foot cliffs of the Gorge.

and careful computation, the total loss of volume was .5 dB on the longest stretch. The wires were covered with neopline without which the wiring would have deteriorated within six to eight months.

When the equipment arrived on the morning of April 12th, the Gorge was ready for the installation. The first day was spent with the construction crew lowering speakers into the Gorge from 110 foot cliffs (see FIGURE 5). The speakers on the show wall, facing the audience, were hauled across the Gorge. The other speakers were lowered from the top of the cliffs, using a tripod extension on a crane. The speakers were mounted on concrete palates that were positioned in January. By noon on the following day, the racks were in the control booth and the system was wired.

"That afternoon we played the audio show for the first time. There were seven channels of audio spread far apart around the Gorge, the sound and sense of motion between the speaker location was fantastic," said Blake.

At one point, while a speaker was being lowered into position, it started spinning and couldn't be controlled. The screen covering the speaker was torn, but was easily replaced.

THE FINAL PRODUCT

The format for the show begins with the meeting and gathering of the audience. In the pavillion/bandstand area

there is wine and cheese available, musical entertainment, and a chance to get acquainted with Timespell through literature, brochures, and their museum. As dusk falls, the show's operator depresses the Audience Walk In button; this brings up the lights and starts the cassette deck. While the background music plays, the audience is walked through

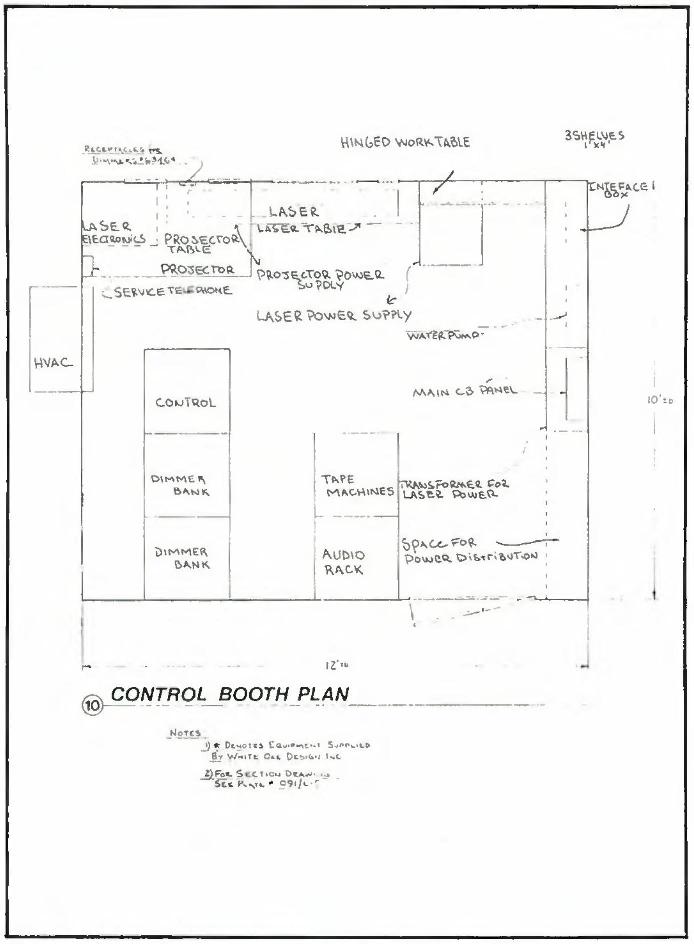


Figure 6. The control booth plan for Timespell sound-and-light show.

a natural stone tunnel, across a bridge, and into the breathtaking Gorge. When the audience is ready, the "Stand by to Announce" button is pushed, fading out the music and allowing the announcer to speak through the system. Using a push to talk microphone, the announcer prepares the audience for the beginning.

Timespell begins with a push of the Start Show button, starting the 8-track. From this point on, the laser, lighting, and audio effects become completely automatic.

At the end of the show, the 8-track rewinds to a foil strip at the beginning and plays until the computer senses time code, stops the 8-track; it's then ready for the next performance. The background music fades in and the audience leaves.

"The outdoor conditions, the rocky precipices and the tumbling waterfalls all presented audio problems of the highest order, and yet the system sounds just gorgeous!" said Jacobsen of White Oak Design. Timespell runs from early spring until the fall, when the software and hardware are removed for the winter months.

Timespell opened right on time and each of the performances has been a sell-out. The Watkins Glen area has prospered with the increased tourism, as hoped for and, of equal importance, the Watkins Glen Gorge is being treated with the respect that it has earned through the centuries. ■

DONNA LEWIS CURTIS

Lasers Light Up Gorge

Image Engineering Corporation, a laser graphics and display company from Somerville, Massachusetts, was commissioned by John Jacobsen of White Oak to design and install a laser projector system technically compatible with all other components of the audio visual system for Timespell.

The laser system, designed by Eric Eisack, was required to project 90-foot-wide graphics onto the gorge wall 120 feet from the projector, and generate atmospheric effects as well.

THE SYSTEM ASSEMBLY

A custom designed and constructed projection system integrating an argon laser capable of producing blue green light was used to create atmospheric beams as well as unusually large representational graphics. The system consists of three major parts: the laser and its electronics; a custom designed optical bench and optical components, and the custom designed image and optical drive electronics.

The laser, a Lexel model 95-4 argon, was modified by Image Engineering to produce as much as 7.5 watts of laser light. The unit requires 208v 3-phase power at 35 amps and water cooling at a rate of approximately two gallons per minute at 20 psi.

The optical bench and components were custom designed by Image Engineering as well. For this application, the laser is integrated into a system of precision optical components allowing for two modes of operation. In beam mode, mechanical actuators can be used to project individual beams onto mirrors to create matrices of light in the gorge. Accumulated beam lengths of approximately 350 feet after two reflections required angular accuracies of approximately ±.75 milli radian! The second mode, scanning, allows for the projection of imagery onto the opposing cliff face.

The image and optical drive electronics were also custom designed for compatibility with the optical components and the signals from the unique control computer. The drive electronics designed by G. B. Engineering allow the control computer to manipulate any element of the Image Engineering system—turning the laser on or off, increasing or decreasing power for a fading effect, selecting the preprogrammed image, or activating the atmospheric beams. The 16 pre-encoded images, stored in the electronics rack, are called up by the show computer one at a time.

THE INSTALLATION

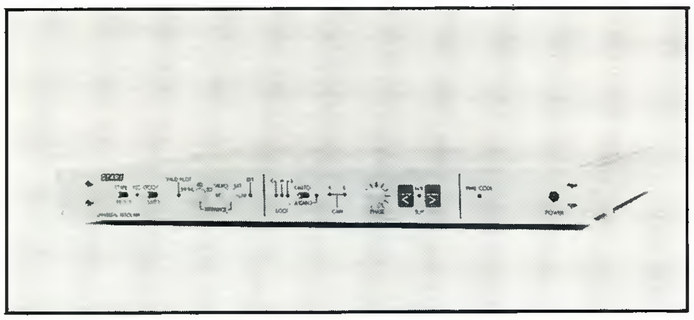
The show computer and the audio and light equipment, including lasers, are housed in a permanently enclosed lighting booth to protect them from nature's elements. IEC master rigger, Bryan Hemberger, rappelled over the edge of 150 foot cliff faces into the gorge by rope and harness to permanently place seven 8- by 10-in. remote mirrors on accurate aiming devices on the rock's surface to produce the complex reflection pattern. Reflecting the beam off these mirrors produces a beam sculpture in the gorge, and the natural mist in the atmosphere enhances the beam, making it more radiant.

In addition to beam sculptures and graphics throughout the show, lasers were creatively employed to produce other special effects such as a 150 to 350-foot beam projected across from the parking lot as a dynamic opening effect, a dramatic lightning bolt, and special images introducing each act. ■

system offers independent mute and level automation accurate to one video frame. Update data is automatically merged with old data to provide a new composite mix. The system includes a bi-directional multi-standard time code generator and can be interfaced to most automation-ready consoles or retrofitted for consoles not previously prepared for automation. The use of time code also permits transport interfacing using the Audio Kinetics Q lock 3.10c synchronizer. The Q lock can provide all of the operating code standards: EBU 25 frames per second, SMPTE 30 frames per second, SMPTE drop frame, 29.97 frames per second, and film 24 frames per second. A 310-3 system provides synchronizer and audio editor, generator, and remote control for three audio or video machines with auto record, autolocate, cycle routines, and 10 user memories. Special software is available for ADR/looping, sound effects assembly, and telecine.

OTARI, SONY, MCI/SONY, dbx

In addition to their rapidly growing line of MTR-series microprocessor-controller tape recorders, Otari exhibited and released preliminary information for the EC-401 and EC-402 resolvers, designed as multi-purpose speed controllers for audio tape machines—especially as used in film and video interface applications. Using a tape machine control track of biphase, FM pilot, SMPTE/EBU, or any 40 to 80 Hz signal, and external reference of SMPTE/EBU, composite video, mains, or TTL level signal, Otari's proprietary Widelock constant phase circuit can lock the machine's control track to an external or internal time base reference. The 402 resolver is a dedicated plug-in module for the



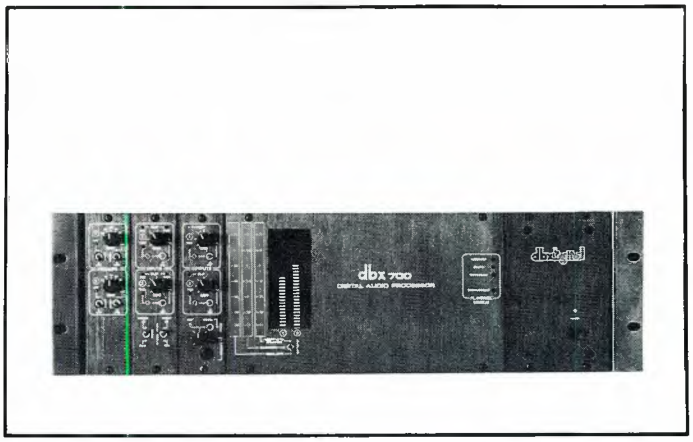
Otari's EC-401 Universal Resolver.

MTR-10/12 tape machines; the 401 universal resolver also includes time code reader/display and capstan control outputs.

Sony and MCI/Sony were present in full force at the convention. Apparently not content with the acquiring of the South Florida console company, Sony announced a new 12 channel portable mixer, the MX-P61, a rack size console with transformerless balanced input and output, monitoring

Sony's RM-3310 Remote Control System for the PCM-3324.

and talkback, three band equalizers, two stereo limiters, and phantom powering. Also demonstrated was the Sony RM-3310 remote controller for the PCM-3324 24-channel digital recorder. The new unit provides full remote control as well as synchronization of additional (up to 15) multitrack recorders, and autolocate functions. The RM-3310 consists of rack-mounted master unit and system control board. Sony also introduced the CDA-5000 Compact Disc analyzer, companion to the CDP-5000 professional disc player. The CDA-5000 can be used for quality assurance or CD masters and replicated discs, checking for sub-code dropout or irregularities, noise generated by the error rate, and mistracking of the laser pickup. The CDP-5000 professional console CD player is intended for broadcast studios, and features quick access capability—search to any audio signal within 13.3 ms accuracy, within an access time of



The dbx Model 700 Digital Audio Processor.

2 seconds. The MCI/Sony JH-110B-3-LB layback system for video was debuted at the convention; this is a post production recorder designed for transfer of audio to 1-inch type C video tape—to be used instead of a type C VTR. The layback recorder has two audio tracks and one SMPTE time code control track with 1-inch tape, and optimizes audio parameters for the audio portion of the edited video tape.

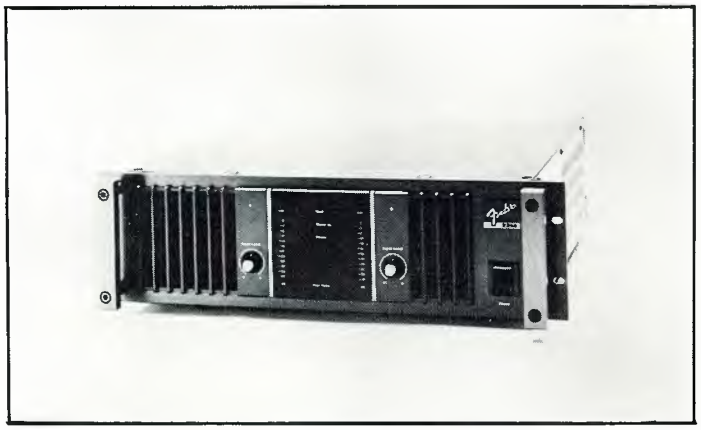
In addition to their line of signal processors and noise reduction units, dbx's answer to linear PCM, the Model 700 digital audio processor, was demonstrated at the convention. Their process uses companded predictive delta modulation, sampling at 700 kbit/sec to obtain a dynamic range of 110 dB, frequency response of 10 Hz to 20 kHz $\pm\frac{1}{2}$ dB and THD of less than 0.01 percent at 1 kHz. The rack-mounted processor uses a conventional video cassette recorder as its storage medium. Delta modulation codes only the differences between successive samples rather than instantaneous voltage values as in PCM. However, delta modulation's dynamic range is limited. In the dbx 700, the limitation is overcome—a compander is used to optimize the signal level to the delta modulator, and a linear prediction filter is used to lower the quantization noise.

QUANTEC, TECHNICS, FENDER, MXR

The Quantec room simulator from Marshall Electronics is "more than just a reverberator." In fact, it is a digital processor that can emulate the acoustical behavior of any volume, establishing a continuum of resonances correct in their density and distribution. Specifically, room size, decay time, low frequency decay, high frequency decay, reverberation density, resonance density, reverberation, and first reflection are all computed and simulated. Thus any conceptually correct acoustical environment may be devised to best suit the musical context—whether it be a recreated acoustic environment, one created anew, or perhaps an environment not naturally acoustically possible. This device

uses a 16 bit A/D converter and 26 bit processor clocked at 20.48 MHz, with 2 Megabit of RAM.

Technics has announced the development of a 16-channel digital audio tape recorder using ¼ inch tape. High density (4.06 Megabits/square inch) recording is accomplished through the use of thin-film heads. In addition to the 16 audio channels, there are three auxiliary channels and one control channel. An isolated loop tape transport is used, and a remote control provides for auto/manual punch-in, autolocate and cycle, pitch control of plus and minus 9.9 percent, 10 point memory of tape position, and 10-group set-up memory. The tape recorder features switchable 48/44.1 kHz sampling rate with 16 bit linear quantization, and tape speed of 30 ips at 48 kHz.



Fender's 2244 Power Ampifier, delivering 440 watts per channel into 8 ohms.

Fender, long known for their expertise in the construction of musical instruments, has expanded their product line to include sound reinforcement equipment. Fender mixers now include models from the 6-channel Model 3106 with built-in power amplifier to the 16 input stereo Model 4216, all with phantom power, transformerless circuitry, and LED indicators on every input channel. Two stereo Fender power amplifiers are offered: the 2224, with 240 watts per side, and the 2244, with 440 watts per side. Both have balanced inputs, selectable high pass filter, forced air cooling, speaker disconnect relays, and LED metering for level and heat overload. Fender microphones include the M series miniature condenser capsules and remote preamplifier, and the D series of hand-held dynamic microphones.

MXR's Model 185 Drum Computer.

MXR has expanded its audio processing line to include a drum computer that features 12 digitally recorded drum sounds with individual level controls and outputs, a capacity of 100 patterns of up to 99 beats each, tempo adjustment from 40 to 250 beats-per-minute, click track, and external trigger for all voices.

NEW ENGLAND DIGITAL, AKG, PHOENIX AUDIO LABORATORY

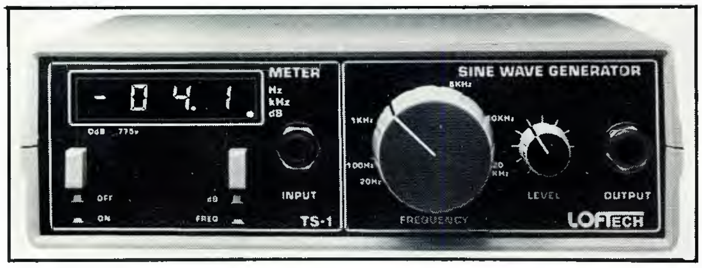
New England Digital has announced new Synclavier II options including a digital guitar option with access to all of the Synclavier's capabilities (synthesis, printing, sample-to-disk etc.). A 16-button LED panel attaches to all Roland GR guitars to control the Synclavier's real-time features.

New England Digital's Synclavier II Real Time Keyboard, part of the Synclavier II Digital Music System.

Just when I hoped that AKG was ready to introduce a digital microphone, they shook me up with a new tube microphone named The Tube with, yes, vacuum tube circuitry. This microphone uses a large diameter condenser capsule and hand-selected 6072 tubes; a remote unit provides for nine pattern selections and three position bass roll-off; it comes in a flight case to protect those fragile pieces of glassware.

AKG's newest invention in microphones, The Tube.

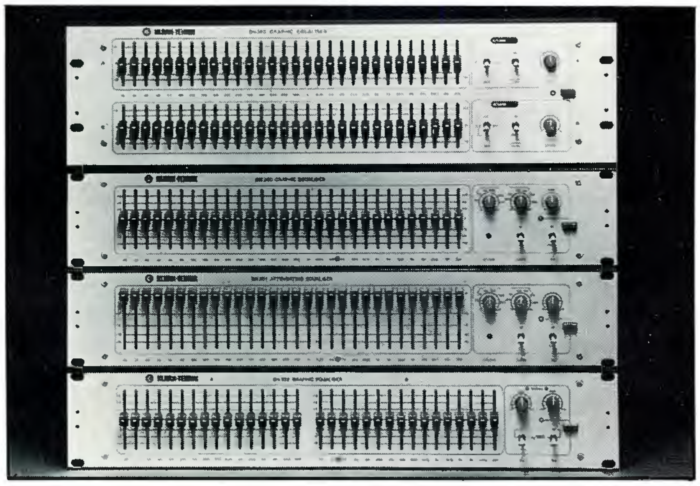
Phoenix Audio Laboratory demonstrated its nifty Loftech TS-1, an audio measurement device that combines a sine wave generator, dB meter, and frequency counter in an almost palm-sized case (your hand, not the tree outside my window). Also included with the test set is a book explaining basic calibration and measurement procedures including S/N ratio, frequency response, resonance and impedance.



Phoenix Audio Lab's Loftech TS-1 Audio Measurement Device.

KLARK-TEKNIK, LEXICON, BEYER, APHEX

Klark-Teknik has introduced a new family of equalizers, building on the reputation of the DN27A equalizer. The DN360 combines two 1/3-octave channels over a full 30 ISO center frequencies from 25 Hz to 20 kHz. The DN301 is a 30-band attenuating equalizer, while the DN332 offers dual 16-channel 2/3-octave equalization.



Klark-Teknik's new family of equalizers.

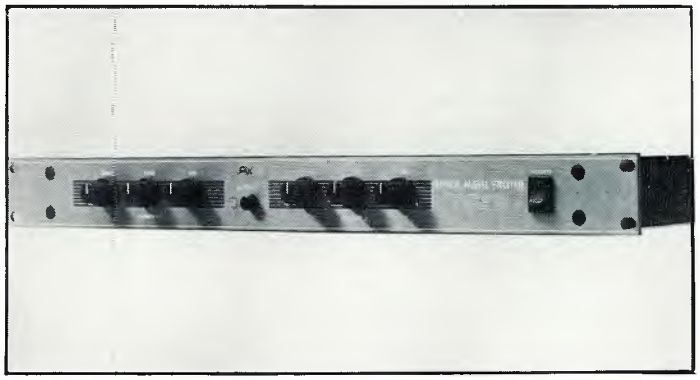
The capabilities of the Lexicon 224X digital reverberator have been expanded with the introduction of the LARC (Lexicon alphanumeric remote console), which displays the selected family of programs, specific program within the family, and parameters within the program; the LARC also permits cassette storage of set-ups. Split levels permit simultaneous use of two reverberations or reverberation with effects. Field conversion to the LARC for existing 224Xs is available.

Lexicon's 224X Digital Reverberator with LARC.

Beyer presented its MPC 50 Acoustical Boundary mic, which utilizes the increase in sound pressure level of acoustically live surfaces but does not pick up reflections from them. Also featured were the MC 736 and 737 Shotgun Condenser mics, both incorporating a 12 dB attenuation switch and a bass roll-off switch to compensate for the proximity effect.

Beyer Dynamic's MPC 50 Acoustical Boundary Mic.

Aphex provided aural excitement for the convention, especially with its Type B aural exciter. Type B has two independent channels in which settings can be matched for stereo effects, a mix control that varies the amount of enhancement fed back into the program, and a tuning control that adjusts the frequency range of the enhancement. The Aphex exciters use a psychoacoustic effect to restore presence, improve intelligibility, and increase perceived loudness without changing actual gain.



The Aphex Type B Aural Exciter.

BTX, SSL, NEVE

BTX demonstrated its Shadow SMPTE synchronizers and Cypher time code system, as well as the new Softouch transport controller. The Cypher can simultaneously read, generate, and character insert SMPTE/EBU code in both longitudinal and VITC formats. An RS 232/422 interface permits added versatility and expansion, while a plug-in relay option provides 16 programmable time code synchronizer contact closures. The Softouch controller interfaces with both the Shadow and Cypher systems and offers multimachine control of features such as record in/out, automatic dialogue replacement, and sound effects editing. The Softouch can store up to 16 production sequences simultaneously and recall them with a keystroke.

Solid State Logic promoted the advantages of sophisticated console control, as found on their 4000 and 6000 series consoles. Both consoles use extensive software programs to extend the flexibility of the mixing desk and reduce logistical overhead. The Total Recall System permits storage of all

BTX's Cypher Time Code System.

module settings, and the Real Time System permits the assembling of rehearsal presets into a performance sequence. All SSL computer programs are stored on a Program floppy disk, and user material such as track lists, cues, set-ups, mixes, presets, sequences, and events are placed on a Production disk. An optional printer may be fitted into the producer's desk for hard copy of mix parameters. There is not, however, a cigarette lighter built into the arm rest.

Neve's DSP, the world's first digital mixing system.

Neve has released a progress report on the development of its digital mixing console. Since the completion of the prototype DSP in 1981, over 300 users have evaluated its performance characteristics and contributed suggestions for its refinement. Presently, two digital consoles for BBC's OB and CTS studios are nearing completion. They feature direct digital interfacing to digital recorders and digital transmission systems, programmability of controls, memory, and automation for all controls including faders, compressors, equalizers, pan pots and routing.

DASH

Finally, one of the major announcements at the convention involved a standard sure to influence coming generations of hardware. A new digital audio stationary head (DASH) standard was announced by Studer, Sony, Matsushita, and MCI/Sony in which the use of thin-film heads, low-speed formats, and the recommendations of the AES Standard Committee on Digital Audio are all provided for. Dash provides a format encompassing 2-channel (¼ inch tape) to 48-channel (½ inch tape), with three tape speeds (7½, 15, 30 ips) that determine the number of coded data tracks. Both sampling rates of 48 kHz and 44.1 kHz are supported by the format at all three speeds. A newly developed modulation code, HDM-1, permits a longer recorded wavelength than that with MFM, the linear packing density is 38.4 kbits/inch, error protection employs Cross Interleave Code, and editing can be accomplished with punch-in, razor blade, or electronic editing.

The Technical Papers

Hidden somewhere amidst the frantic marketplace of exhibitors, the blurred vision of the hospitality suites, and the temptations of New York City nightlife, was the raison d'etre for the convention—the opportunity to serve the Society's purpose of education and exchange of information concerning audio technology, in both theory and practice. Almost eighty technical papers (representing the work of over one hundred authors), nine workshops, special presentations and a tutorial representing over fifty speakers provided substantial insight into the state of audio recording. This 74th convention was exceptional in its success in combining the too often irreconcilable tasks of teaching and scientific discourse. It is to the Society's great credit that many professionals, as well as many students, were able to share in both endeavors. The range of topics and their levels of presentation was ambitiously planned and superbly executed. A look at the convention calendar should excite and fatigue even a casual observer, and bring restless reminiscences to those who participated in it.

SATURDAY
Technical Session—Digital Recording and Broadcasting
Workshop—Hands-On Digital
Tutorial—Digital Audio Topics
Presentation—Compact Disc
SUNDAY
Technical Session—Analog Signal Processing
Technical Session—Studio Design
Workshop—Economics of the Recording Studio
Workshop—Microphone Techniques for Stereo Video

MONDAY
Technical Session—Sound Reinforcement
Technical Session—Loudspeakers: Low Frequency Alignment
Technical Session—Loudspeakers: Network Considerations
Workshop—SMPTE Code and Synchronization
Workshop—Console Troubleshooting

TUESDAY
Technical Session—Digital Signal Processing
Technical Session—Psychoacoustics and Subjective Testing
Workshop—Audio Production for Motion Pictures
Workshop—Multitrack Tape Machine Maintenance

WEDNESDAY
Technical Session—Test and Measurement
Technical Session—Disk Recording and Multichannel Sound
Workshop—Digital Recording Techniques
Workshop—Grounding and Shielding

The execution of this calendar required five days and the efforts of 150 authors and speakers; even the set of preprints available from the AES represents a year of challenging reading. In other words, any summary must be hopelessly incomplete. Nevertheless, I would like to document the topics which the technical papers covered and point out a few of the presentations which intrigued me or otherwise commanded my attention.

DIGITAL RECORDING AND BROADCASTING

The Digital Recording and Broadcasting session was chaired by Robert Adams of dbx and included papers concerning the use of thin film heads for digital recorders, channel coding for stationary head PCM, digital audio modulation using the PAL and NTSC formats, the ECC error code correction, labels and formatting, delta modulation, the digital audio stationary head DASH format, digital audio transmission via satellite distribution, Reed Solomon decoding with multiple passes, and a report of work on a digital audio compact cassette recorder.

The paper on thin film head design was authored by Yasuharu Shimeki, Misao Kato, Shiro Tsuji, and Hiroshi Matsushima of Matsushita Electric; it described the development of a 16 channel digital tape recorder employing ¼ inch tape. A thin film head permits this track width reduction and offers the following advantages: reduced power consumption, 20 dB reduction in head crosstalk, and reduction of peak shift using short wavelengths. A thin film head is constructed on a magnetically grooved substrate; the recording head uses a coil wound around the substrate and the groove is filled with a nonmagnetic material to reduce magnetic leak and improve efficiency. The reproduction head uses the MR magnetoresistive effect of ferromagnetic thin film. For linearity, magnetic unisotropy is imparted with a hairline formed on the MR film, thus providing a reduction in gap width and permitting shorter wavelengths.

Roger Lagadec of Studer and G. McNally of the BBC presented a paper on labels and their formatting. They outlined a format for a recordable and transmittable label track compatible with the AES/EBU serial digital audio interface. Labels could carry various kinds of information: operational data such as program duration, take number, and date; technical data such as wordlength, compression settings, and level settings; commercial data such as copyright information, and additional data such as performers and editorial. They proposed a label format of one label of 48 bits every millisecond, with a length of 192 sampling intervals per block and a 48 kHz sampling frequency, thus providing a transmission rate of 1000 labels of 48 bits per channel per second.

K. Murai, K. Iwakuni, and T. Kogure of Matsushita Electric presented a paper on new signal processing for a digital audio cassette tape recorder. Following a discussion of the design criteria of such a system, they presented the specifications of a prototype digital compact cassette recorder with sampling rate of 44.1 kHz, 16 bit linear quantization, and triple parity error correction. Progress in vertical magnetic recording, thin film heads, and semiconductor technology should permit realization of this recorder in the near future.

The Digital Audio Stationary Head DASH format was introduced by Kogure of Matsushita, Toshi Doi of Sony and Lagadec of Studer. This format utilizes 16 bit audio samples recorded on dedicated tracks at 48 or 44.1 kHz sampling rate at speeds of 30, 15 and 7 ½ ips, with the number of tracks being 2, 4, 8, 16, 32, 48 on ¼ or ½ inch tape with multiple recording densities. Error protection, auxiliary tracks, etc. are all provided for. Embraced by three leading manufacturers, this format might provide the compatibility sorely needed in the digital recorder market. Just as 50 kHz succumbed to 48 kHz, will DASH prevail?

ANALOG SIGNAL PROCESSING

The Analog Signal Processing session was chaired by Robert Cordell of Bell Laboratories and included papers on audio level metering, active crossover designs, audiooptics, power amplifier output stage design, an operational voltage controlled (VCA) element, use of Walsh functions in an FM stereo demodulator, and sound system equalization.

The paper on audiooptics by George Bowley of Audiooptics Technologies provided a glimpse into the possible future of audio; audiooptics integrates optical and fiberoptics sciences with electronics-based engineering to reduce external interference, ground loops, electrical shock hazard, frequency response constraints, nonlinearity, and signal attenuation in transmission. Fiberoptics is taking a primary role in communications and sensing and signal processing applications; the author proposes that audio stands to gain much from this emerging technology.

STUDIO DESIGN

The Studio Design precis/poster session was chaired by Daniel Queen of Daniel Queen Associates and presented discussions on architectural acoustics and artificial reverberation, fluctuation statistics on the source/detection path, control room design, the Schroeder quadratic-residue diffusor, surround sound monitoring environments, design of a multi-format radio broadcast studio, tonal effects of classical music microphone placement, stereo/monaural compatibility, use of boundary-layer-effect microphones, and split track recording for ENG audio.

Greg Badger of Audiometric Laboratories and Chips Davis of LEDE Designs authored a paper on surround sound monitoring environments; they argued that TEF and the resulting improved understanding of psychoacoustics will permit a more sophisticated design of monitoring rooms. Both original soundfield via coincident microphone techniques and multitrack playback via panning could be successfully mixed in a properly designed room.

Skip Pizzi from National Public Radio discussed split track recording in which two audio tracks gather synchronized sound from two independent sources simultaneously; for example, two microphone interviews, language translation or ambience recordings that are later mixed to mono. At the cost of longer post production times, Mr. Pizzi argued for utilization of the multitrack capabilities inherent in the technique to achieve technically superior results.

SOUND REINFORCEMENT

The Sound Reinforcement session was chaired by Eugene Patronis of Georgia Institute of Technology and included papers on TEF techniques, effect of reinforcement system regeneration on gain and reverberation decay, computer-aided central loudspeaker array design, digital control of

array directivity, intelligibility modeling of a loudspeaker cluster, an improved colinear array, a constant-directivity defined-coverage horn, direction-sensitive gating, and a report on good old-fashioned passive acoustics.

Don Davis of Synergetic Audio Concepts outlined the Time Energy Frequency (TEF) concept of measurements of sound systems, stressing the orderly approach in measurements that TEF allows. Time domain is examined via ETC measurements; then frequency domain is examined with Nyquist phase plot, based on the results of the ETC. In addition, TEF perhaps holds the promise of identifying new transforms and new ways to view existing transforms. Implicitly suggested in the paper was that the Crown TEF analyzer is the largest back-ordered device since the IBM PC.

George Augspurger of Perception, Inc. and James Brawley of James Brawley Associates authored a paper on an improved colinear array for sound reinforcement. In the past, problems with loudspeaker line arrays have included pattern control, sound power and cost. In this new three-way design all transducers are positioned side by side to obtain better response in the horizontal plane and suppression of side lobes. The authors confessed to using 8 bit personal computer simulation to facilitate their work; acousticians are a lazy lot, but honest.

LOUDSPEAKERS

The Loudspeaker session on Transducers and Low Frequency System Alignments was chaired by John Eargle of JBL and JME Associates, and included papers on impulse testing of loudspeakers, amplitude of frequency modulation distortion, measurement of spatial reproduction quality with interaural cross correlation, optimizing for wide area imaging, a dual moving coil transducer, nonlinear distortion reduction, vented box design using root locus, and a high efficiency servo driven subwoofer.

Laurie Fincham of KEF delivered a paper discussing refinements in the impulse testing of loudspeakers. The author summarized the use of digital processing of a loudspeaker's impulse response to evaluate its frequency response, and detailed sources of measurement effort particularly at low frequencies. The author proposes a new method for measuring low frequency response, utilizing spectral shaping of the test signal to suit the unit under test, and digital post-production of the collected data—resulting in an accuracy of better than 1 dB from 20 Hz to 50 kHz with a measuring time of one minute in a nonanechoic environment.

The Loudspeaker session on Networks was also chaired by John Eargle and presented papers on passive three-way constant power and all pass crossovers, elimination of lobing error in multiway speakers, synthesized three-way crossovers, influence of crossover design on the power response of speakers with noncoincidental drivers, a discussion of the complexity of the design philosophy for crossovers, and in-phase crossover design.

Eugene Zaustinsky of SUNY at Stony Brook delivered a paper with a question: Is sophisticated loudspeaker crossover design possible without sophisticated measurements? Specifically, can classical design approaches dealing with measurement of acoustic phase be avoided in favor of a design algorithm based solely on amplitude response measurements? Zaustinsky proposes a method that produces a crossover design limited only by driver response defects and enclosure diffraction effects. Knowledge of the location of the acoustic centers of the drivers facilitates the technique.

DIGITAL SIGNAL PROCESSING

The Digital Signal Processing session was chaired by James Moorer of Lucasfilm and included papers on a single stage sampling frequency converter, digital studio signal transmission and routing, a consumer auditorium simulator, a digital reverberator with early reflection control, design of a linear-phase digital equalizer, signal enhancement with digital signal processing, a digital audio console, and a digital audio control center.

Lagadec, D. Pelloni, and A. Koch of Studer presented a paper on a single stage sampling frequency converter—an improvement on the previous four stage digital filter converter. The new model utilizes one filter of length 1.8 million, can reverse modes without output discontinuity, and processes word lengths over 16 bits. Computation of coefficients in real time is aided by correlation between neighboring coefficients and efficiency of computation is increased with the use of an external processor.

Jeffrey Borish of Signal Research Laboratory authored a paper that described an auditorium simulator for home use. Early reflections that are normally excluded from stereo recordings are synthesized by the processor to augment the ambient information already contained in the recording. Two additional loudspeakers with noncritical placement are required.

Lagadec and Pelloni of Studer authored a paper on signal enhancement via digital signal processing in which a digitized audio signal is split into phase linear, complementary, narrow band signals for real time, time domain processing with enhancement algorithms. Recorded examples illustrated the tremendous potential of such a system, and the need for continued research on intelligent (perhaps artificially intelligent?) signal enhancement algorithms to differentiate between signal and unwanted signal.

PSYCHOACOUSTICS AND SUBJECTIVE TESTING

The Psychoacoustics and Subjective Testing session was chaired by Floyd Toole of the Canadian National Research Council and presented papers on the subjective importance of low frequency group delay, audible effects of time delays, phase distortion in anti-aliasing filters, perceived sound quality of high fidelity loudspeakers, subjective measurements of loudspeakers in stereo and monaural listening, measuring methods for headphones, evaluation of localization in surround sound, and a subjective comparison of five analog and digital tape recorders.

D. Preis and P. J. Bloom of the Electrical Engineering Department of Tufts University authored a paper on the perception of phase distortion of anti-aliasing filters using minimum phase 4 kHz and 15 kHz filters. Tests using broadband clicks showed that a cascade of up to four pairs of seventh order elliptic filters introduced no perceptible effects, with diotic presentation via headphones. Other tests using speech and music via loudspeakers await completion.

Martin E. G. Willcocks of Willcocks Associates and Greg Badger of Audiometrics authored a paper on localization and psychoacoustics in surround sound; extension of stereo to more than two loudspeakers has required a rethinking of localization. The authors proposed an extension of Mertens's theory to take into account multi-loudspeakers and localization cues for non-optimum speaker placement in which pairwise mixed discrete surround sound offers the best results.

Wieslaw R. Woszczyk of McGill University and Floyd M. Toole of the Canadian National Research Council authored a paper describing a comparison of two analog and three digital tape machines using double blind tests. Twelve listeners provided analytical ratings. Results showed that differences between machines were extremely small and, in fact, the differences between analog and digital were not readily distinguishable. Perhaps both analog and digital manufacturers breathed sighs of relief at this finding.

TEST AND MEASUREMENT

The Test and Measurement session was chaired by W. Marshall Leach, Jr. of Georgia Institute of Technology and included papers on the design of programmable distortion measurement systems, measurement of the distortion of frequency modulators and demodulators, measurement of digital audio system phase linearity, microcomputer post-processing for FFT analyzers, phonograph cartridge

evaluation with minishaker and TDS, and the HP-41 as sound level measurement tool.

a sound level measurement tool.

Lagadec of Studer, who won the gold medal for trips to the speaker's rostrum, delivered a paper on the measure-the use of "square wave symmetry" as a criterion for linear phase response, being independent of amplitude response. A simple measurement of phase non-linearity is the amount of asymmetry in the response to a symmetrical square wave. Hardware involves only low-cost electronics. Phase compensation is also made measurable by the new figure of merit.

DISK RECORDING AND MULTICHANNEL SOUND

The Disk Recording and Multichannel Sound session was chaired by Duane H. Cooper. Papers were presented on ambisonics, surround sound decoders, models for record groove skipping, mastering techniques for test records, a perspective of stylus/groove interface, and a history of lateral tracking geometry and its effect on playback.

Michael Gerzon, consultant to National Research Development Corporation, presented a paper on ambisonics in multichannel broadcasting and video; the ambisonic UHJ encoding hierarchy for transmitting sound fields offers more mutually compatible upward applications to any number of loudspeakers and permits options such as periphonic reproduction. It was pointed out that ambisonics encompassed quadraphonic systems and permits both broadcaster and listener to make their own approximation to the recreation of the original sound field with great compatibility to future applications.

WHAT IT ALL MEANS

I think this glance at the papers presented at the 74th convention confirms the fact that the science of audio recording and reproduction is operating in a healthy environment, characterized by new ideas and innovation. Analog techniques have achieved a zenith of fidelity and challenge the present limits of digital audio's new technology. Clearly both technologies will compete for years to come, with analog reluctantly stepping aside only when digital has proved itself against analog's high standards. The audio community has also turned its attention to other facets of the science—surround sound and the importance of more sophisticated image placement and the creation of artificial acoustical environments have surfaced as new terrain to explore. Thanks to audio researchers such as those present at the AES convention, the sound of recorded music promises to get better and better. ■

New Products

WIRELESS AUDIO LINK

• The Precision Audio Link (PAL) system gives flexibility to acoustic engineers by providing a precision wireless link between sound measurement points and analysis and recording instrumentation. It is intended for remote acoustic measurements, sound contractor data analysis, sound reinforcement installation measurements and sound system maintenance. The audio bandwidth—20 Hz to 20 kHz—coupled with 100 dB dynamic range and 5 ppm frequency tolerance, results in complete transparancy of the audio signal. The PAL system is compatible with TEF analyzers and spectral analysis equipment, as well as most sound level meters and calibrated microphones. The PAL system was developed with the support of Don Davis of Syn-Aud-Con.
Mfr: HM Electronics, Inc.
Price: $2,200.00
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MICROPHONE ACCESSORIES

• Beyer Dynamic offers a complete line of microphone floorstands and booms to complement their microphones. The ST 201 A/1 microphone stand features height adjustment ranging from 36 to 64 inches and can be used with a boom. It also includes 13-inch heavy-duty screw-in legs, a two section telescopic column, and a noiseless screw-type locking device. The ST 210 A/1 mic stand includes the ST 201 A/1 floor stand and the SCH 211 lightweight boom arm, which extends to a maximum length of 30 inches and can be pivoted through 360 degrees in any two planes. The ST 201 A/2 mic stand is similar to the ST 201 A/1 but features 13-inch heavy-duty folding legs and a sliding base. The ST 210 A/2 mic stand includes the ST 201 A/2 floor stand and the SCH 211 lightweight boom. The ST 259 is a low-profile microphone stand with a noiseless screw-type locking device and 10-inch fold-away tubular legs. The Beyer ZMS 2 is a microphone bar for the mounting of two microphones on one floor stand. The maximum spacing between the microphone fixing points is 7.8 inches, making the ZMS 2 ideal for mic'ing rack toms.
Mfr: Beyer Dynamic

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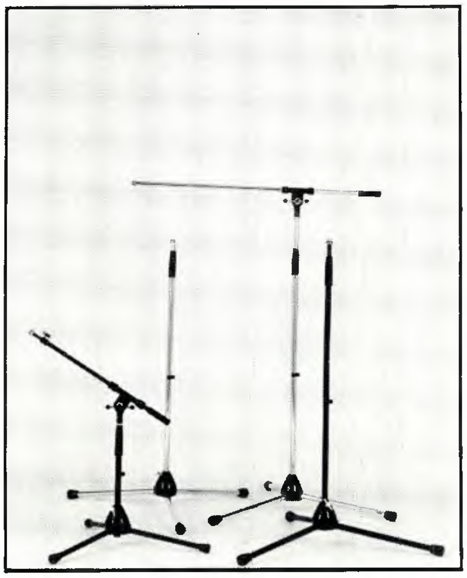

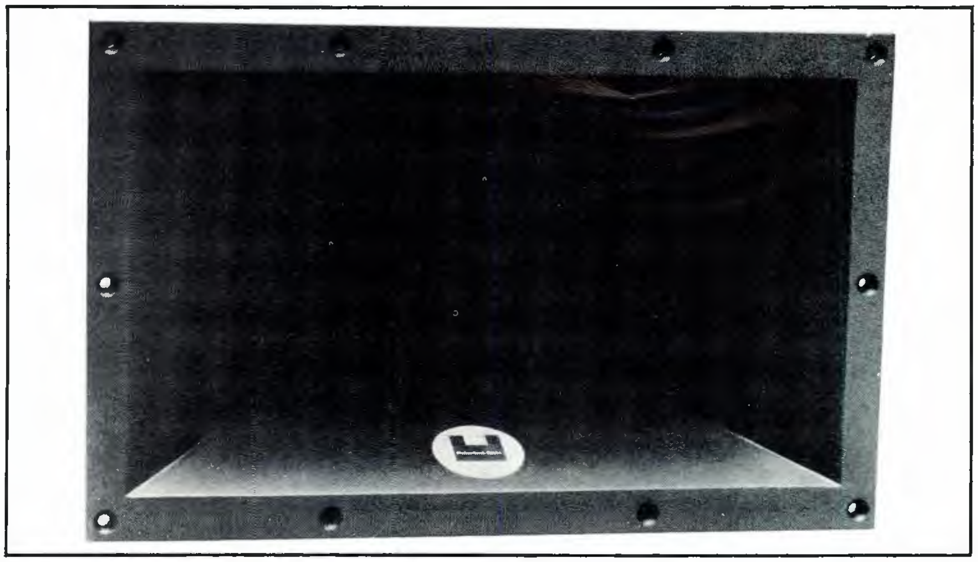

• The Model 2386 Bi-Radial Constant Coverage Horn provides uniform on- and off-axis frequency response (from 500 Hz to 16 kHz in the vertical plane), with a nominal 40 × 20 degree coverage angle. The 2386 has been designed for flush cabinet mounting or compact cluster installations. The performance of the new flat-front 2386 simplifies cluster design, minimizes the need for horn overlapping, and virtually eliminates lobing and comb filter effects. Computer-aided techniques were employed to derive the horn contours in the horizontal and vertical planes; this design yields smooth response, low distortion, and even coverage, avoiding the performance disadvantages of sharp flare transitions and flat sidewalls. The 2386 is constructed of injection-molded reinforced polyurethane. The horn's small vertical mouth dimension (slightly larger than the compression driver used to drive the horn) allows compact single and multiple horn/driver systems to be assembled. Should vertical pattern control be needed below 2 kHz, two or more horns may be stacked vertically to restore full Bi-Radial performance. The 2386 will accept JBL's 2-inch diameter 2441, 2445 or 2482 compression drivers. With the addition of the 2327 adaptor, the horn also will accept the 1-inch throat 2525 driver.
Mfr: JBL

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NOISE REDUCTION SYSTEM

• Rocktron Corporation's System One encode/decode noise reduction is an advanced system in an unbalanced format. Available in two, four, or eight channels, it is designed to be used with high speed recording equipment (15 ips, 30 ips) that is capable of providing flat frequency response (±1 dB from 30 Hz to 20 kHz). System One's innovative design takes advantage of today's improved magnetic tape. It effectively doubles dynamic range by compressing the dynamic range onto the tape during the recording process and expanding it upon playback. This allows the user to record the full dynamic range of the source. The System One greatly improves recording capabilities while decreasing distortion caused by tape saturation and eliminating tape hiss. It provides effective noise reduction greater than 40 dB, and is claimed to be the first companding noise reduction system to eliminate the obtrusive side effects of modulation noise (breathing), and pumping. System One's design, using high quality components and tightly matched encode/decode response, provides a transparent system without coloration. In addition, System One operates over the entire audio range offering effective noise reduction at all frequencies, requires no level matching between encode and decode (which allows easy set-up and operation), is fully simultaneous (no switching) and capable of off-tape monitoring of separate channels, and is warrantied for three years on parts and labor.
Mfr: Rocktron Corporation, Inc.
Price: Model 120 (2 channels) $319.00
Model 140 (4 channels) $469.00
Model 180 (8 channels) $829.00

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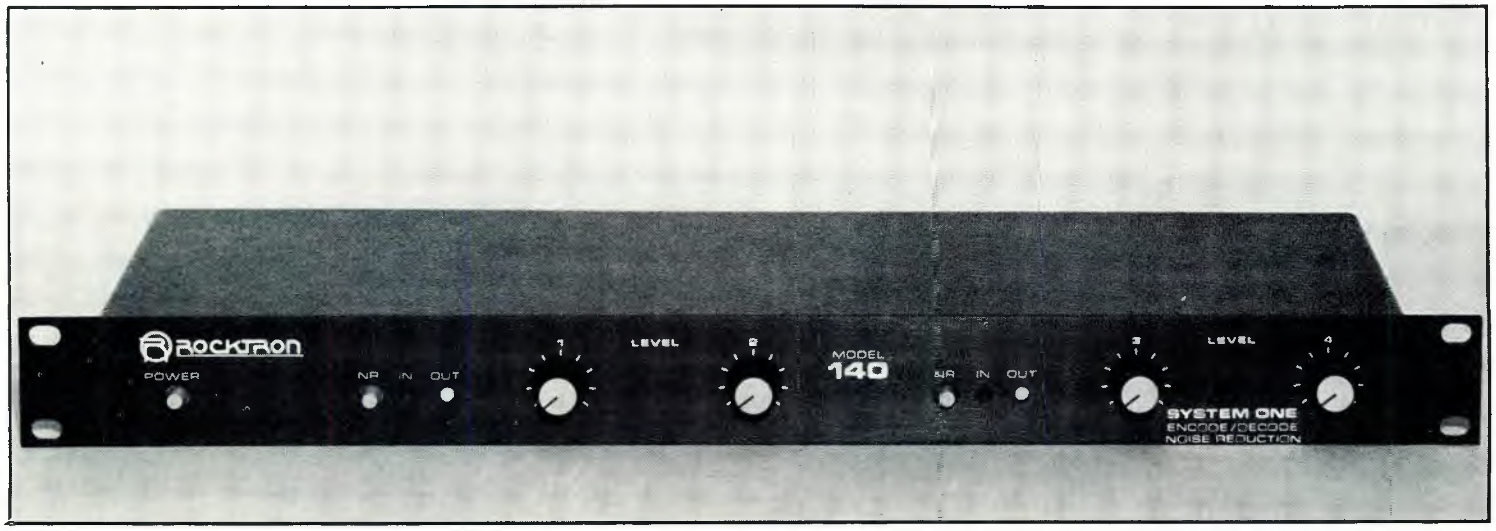

ISO-BALANCED PREAMP

• JBA Research LTD's new "Iso-balanced" preamp, the JBA-207, uses a discreet bipolar transistor design which provides superior audio performance in conjunction with excellent isolation—without the use of transformer coupling. The preamplifier is specifically designed to interface any musical instrument with an audio mixing console. The "iso-balanced" design surpasses conventional balanced systems, providing up to 50 dB greater rejection of ground-loop noise. This eliminates the need for any type of transformer coupling. The JBA-207 Iso-balanced Preamp has two outputs: a high-impedance ¼-inch phone plug output and a low-impedance XLR iso-balanced output. The preamp may be powered by either a 9-volt battery or phantom power. Several other features include a ground-lift switch, ⅛-inch-thick aluminum construction, an input impedance greater than one megohm handling a 3-volt peak-to-peak input signal, and THD less than 0.01 percent. There is unity gain at the high-impedance output and 10 dB of gain into 600 ohms at the low-impedance output. Current users of the JBA-207 include Leon Redbone, Arlo Guthrie, and Fantasy Studios.

Mfr: JBA Research, LTD

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WIRELESS MICROPHONE RECEIVER

• Cetec Vega's new Model R-42 Pro Plus Wireless Microphone Receiver features "infinite gain" technology, ultra-low noise, true dual-receiver diversity, and switch-selectable Dynex® II, an advanced new audio processing technique. System dynamic range with Dynex II is typically 108 dB, A-weighted (maximum deviation to noise floor). With Dynex II switched out, the ULNR (ultra-low noise receiver) has a 92 dB (A-weighted) signal-to-noise ratio. Highest adjacent-channel rejection is achieved with 16 poles of IF filtering. The preselector is a true four-pole helical resonator filter (silver-plated). "Infinite gain" technology provides the best possible signal-to-noise ratio at low signal levels and improved processing of multi-path RF signals. Mu-metal shielding for the power transformer and other critical circuitry eliminates hum and powerline noise. The R-42 diversity (or R-41 non-diversity) Pro Plus receiver provides excellent system performance when combined with the 77 Dynex II bodypack transmitter or the Model 80, 81, or 82 hand-held transmitter.

Mfr: Cetec Vega

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CABLE SYSTEMS

• Canare Cable, Inc. manufactures a full line of bulk and pre-wired cables, junction boxes, pigtails, multi-pair cables, and multi-pin cable assemblies that carry high quality signals without degradation or significant loss. Of special interest is Canare's L4E Series Star-Quad audio cable. This four-conductor cable consists of two twisted pairs plus a high-density braided shield. Because each conductor in the balanced cable actually consists of a twisted pair, the included area between conductors is minimized, maximizing the rejection of AC hum and all forms of electromagnetically induced noise. According to Canare, the resulting noise immunity is 10 times better than standard balanced mic cables. Special cable types are offered to suit the specific technical interface requirements of mic and line level audio, video and musical instrument circuitry. For better identification and visual appeal, guitar cables are available in five colors and microphone cables in ten colors. Cable storage and transportation are simplified with Canare cable reels and cable reel systems. The reels are available for single cables or large multi-pair cables, with or without connector panels. Several models come with three-position brakes that regulate tension or lock the reel completely. Most reels can be stacked for storage and easy pulling of long lines, and some models have roll-around casters. Canare also offers a variety of low-crosstalk multi-pair cables for construction of custom "snakes." Factory-wired 8- to 32-channel snakes are also available with multi-pin connectors that mate either to junction boxes or XLR pigtails. Special combinations of optimized audio, video, and communications cable in one multi-conductor bundle come in five common remote truck formats.

Mfr: Canare Cable, Inc.

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TWO NEW MICROPHONES

• Shure Brothers, Inc. has added two new models to its popular PE (Professional Entertainer) series of moderately priced microphones. The two new models, PE86 and PE66, are unidirectional (cardioid), dual low-impedance models with shock-mounted cartridges for quiet, reliable operation. The PE86's frequency response is 50 to 15,000 Hz; the PE66's is 40 to 15,000 Hz. Both models feature a fixed bass roll-off and the upper midrange presence peak that provides the well-known "Shure sound." The PE86 also features a built-in spherical windscreen to minimize wind and breath noise. The PE86 and PE66 are each packaged with a professional swivel adapter and a vinyl gig bag.

Mfr: Shure Brothers, Inc.

Price: PE86L-LC: $125.00

PE66L-LC: $109.25

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PROFESSIONAL INTERFACE UNIT

• Audio + Design's PROPAK™ Professional Interface Unit lets audio engineers make use of semi-professional and even domestic hi-fi equipment such as digital (EIAJ) video recorders, Compact Disc players, and cassette machines. PROPAK I is designed to facilitate a professional interface for the abovementioned equipment. At the "pro" end of the unit, XLR connectors are provided and gain can be adjusted for any professional operating level. Inputs and outputs are electronically balanced with +24 dBm headroom. On the "semi-pro" end, gold-plated phono sockets are used. Send and Return levels are adjusted on pre-sets to meet whatever operating levels are required, and the unit is AC powered. PROPAK II with CTC™ (Coincident Time Correction) includes all of the above and is also switchable for use with digital recorders operating to the EIAJ standard (eg., Sony PCM F1) to produce a time coincident digital stereo track. Normally, with the EIAJ format, the inputs and outputs are multiplexed so that the recording has a time delay between channels. When reproduced through the D/A, this is corrected. However, taking a direct digital output means that tapes recorded this way are not 1610 or Compact Disc compatible because the signals are not time coincident. PROPAK II provides a cost effective solution while maintaining playback compatibility with the PCM F1. Distortion is less than .003 percent at 1 kHz; dynamic range is 125 dB; and frequency response is 0 to −0.25 dB (20 Hz to 20 kHz).

Mfr: Audio + Design
Price: PROPAK I: $240.00
PROPAK II: $290.00

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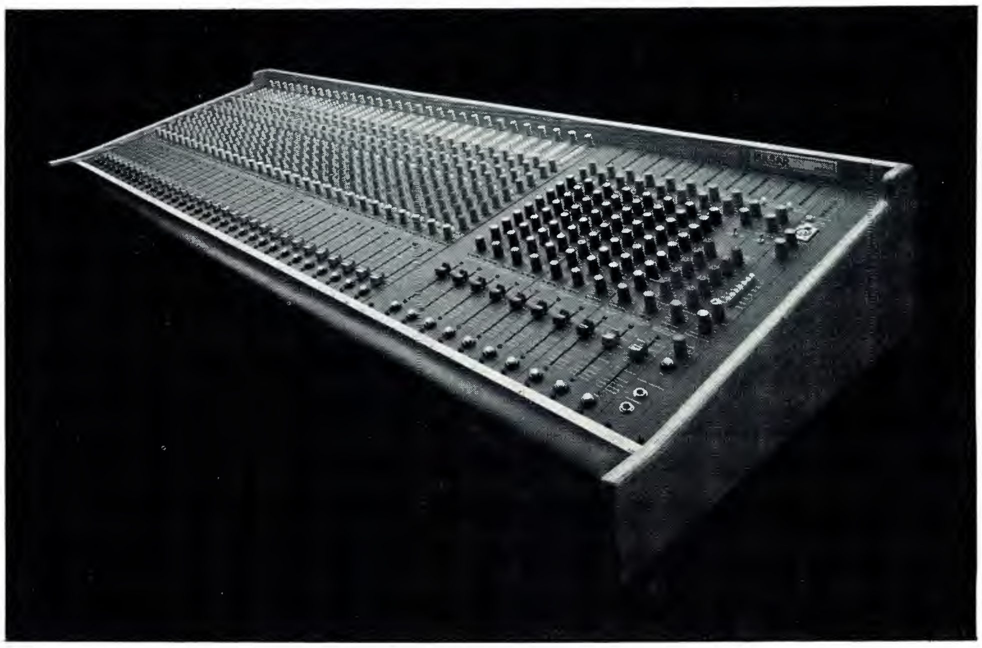
MATRIX MIXING CONSOLES



• Pulsar Labs' 80 and 40 Series Matrix Mixing Consoles are totally modular, allowing the user the option of adding signal processing modules such as four-channel compressor/limiters, dual 10-band equalizers, reverbs, and dual stereo mixdown modules. Full patching is accomplished by access in and out jacks on all modules. Both consoles are suited for live as well as studio recording. Pulsar 80 and 40 Series boards have flexible matrix mixing capabilities which allow eight independent mixes of eight separate groups. All Pulsar mixing consoles come with a solid oak frame with steel sub-frame and a three-year warranty.

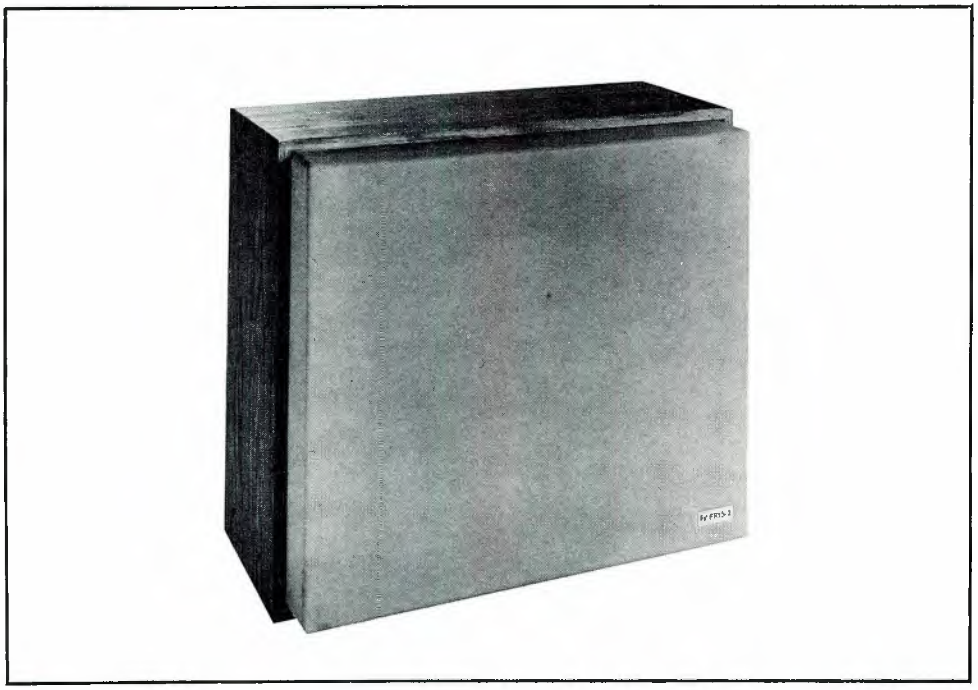
Mfr: Pulsar Labs, Inc.

Circle 57 on Reader Service Card

TWO-WAY SPEAKER SYSTEM

• Electro-Voice's new two-way speaker system, the FR15-2, is intended for sound reinforcement applications where a wide, controlled coverage angle and high efficiency are desired. Low frequencies are handled by a rugged 15-inch EVM-15L Series II woofer mounted in a 4.3-cubic foot optimally vented enclosure. Frequencies above the 15 kHz crossover point are handled by a compression driver on a 90 degrees by 40 degrees constant-directivity horn. Unlike conventional horns, EV's constant-directivity horn maintains its rated beamwidth (90 degrees horizonal, 40 degrees vertical) to the highest frequencies, assuring broad, uniform coverage. The frequency response of the FR15-2 is essentially flat from 50 Hz to 15 kHz. The speaker's sensitivity is rated at 98 dB SPL with a 1-watt input, measured at 1 meter on axis. The long-term power-handling capacity is 200 watts, measured using shaped pink noise with a 6 dB crest factor. The FR15-2 weighs 94 lbs. and measures 28⅜-in. high by 31½-in. wide by 16⅝-in. deep.

Mfr: Electro-Voice
Circle 58 on Reader Service Card



STUDIO RIBBON MIC

• Audio Engineering Associates' Coles 4038 studio ribbon microphone is the culmination of 30 years of development and usage by such organizations as the BBC. The ribbon material is a carefully selected pure aluminum foil which is corrugated and precisely tensioned between highly permeable pole pieces. A slightly magnetic woven screen mounted on each side of the ribbon gives a precise degree of damping and acts to prevent over-stressing of the ribbon under accidental wind-blasts. The ribbon itself combines the functions of a very low mass, critically damped acoustical diaphragm and a low resistance half-turn dynamic coil. The ribbon and pole piece shapes have been optimized for flat frequency response with a higher output efficiency than earlier models. The moving mass of the ribbon is only about 1/500th that of most dynamic moving coil microphone systems, and is easily tuned to a very low basic frequency. All of these features give the microphone a very flat and extended bass response. All microphones tend to suffer a change in the directional (polar) response at high frequencies due to their physical size in comparison to the sound wavelength. The 4038 ribbon and pole piece system is completely symmetrical in the horizontal plane. Combining this with the short vertical length of the ribbon and the acoustically compensating design of the microphone casing helps to maintain a constant shape of the polar response of both planes. The low impedance connections to the 4038 ribbon are made by means of a symmetrical cage of bars dispersed about the ribbon, forming a "humbucking" system giving a high degree of electromagnetic rejection. This low impedance system is substantially earthed against electrostatic interference, and the small toroidal matching transformer feeding to the line is effectively shielded.

Mfr: Audio Engineering Associates
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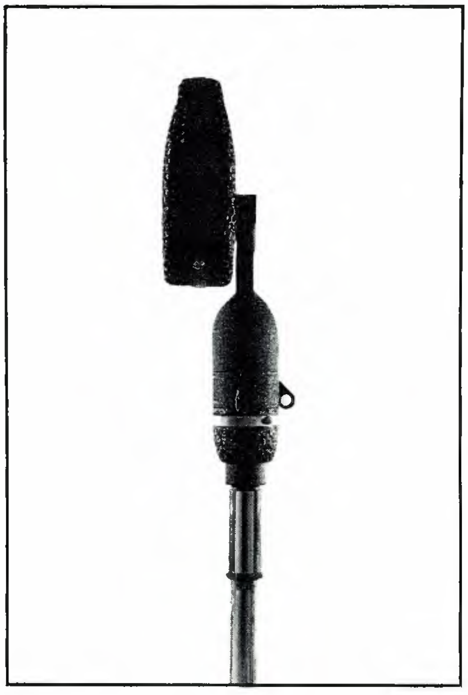


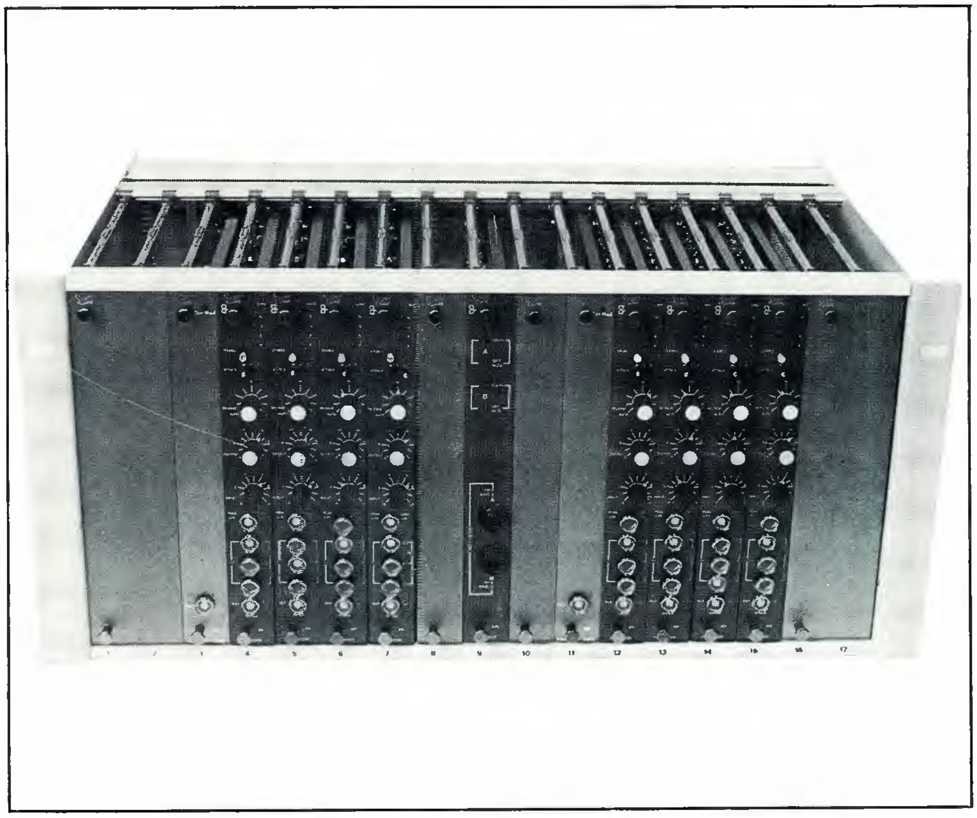
PRO SOUND WOOFER

• The DL15X 15-inch, 8-ohm low frequency driver is designed for high-level monitoring and sound reinforcement. The high-performance drive system of the DL15X is augmented by two Electro-Voice features: the Thermal Inductive Ring (TIR)™ and the PROTEF™ coating. The TIR provides a significant heat transfer path from the top of the voice coil while simultaneously moderating the normal inductive rise of the voice coil. PROTEF, a Teflon-based protective coating, is applied to the inside diameter of the top plate adjacent to the voice coil. It provides protection by lubricating any rubbing contact and inserting electrical insulation between the coil and the top plate. The specifications of the DL15X include a frequency response of 45 to 3200 Hz ±3 dB (in a 6.4-cubic foot enclosure tuned to 42 Hz); a 500 watt long-term-average power capacity per AES Recommended Practice (100 to 1000 Hz), and a sensitivity at 1 meter, 1 watt (average from 200 to 4000 Hz) of 102 dB. EV offers the DL15X with a choice of computer-optimized enclosures for various applications. The 3.2-cubic foot TL606 enclosure with a low-frequency 3-dB-down point (f_3) of 63 Hz is available for those applications demanding a compact enclosure. Alternatively, the 6.4-cubic foot enclosure has a low-frequency cutoff of 45 Hz. In their step-down modes, with appropriate equalization, the f_3's of the two enclosures are 45 and 34 Hz respectively.

Mfr: Electro-Voice, Inc.
Price: $250.00

Circle 60 on Reader Service Card

MULTI-BAND PROCESSOR



• Audio + Design's S27 4-Band Crossover/Summing Amplifier allows the SCAMP Card Modular System to be configured as a cost-effective high quality multi-band processing system for loading the final medium, whether it be broadcast transmitters, tape, disc, Compact Disc, or sound reinforcement. The basic building blocks of the processor are the S27 Crossover/4-Band Processor Module and four SO1 Compressor/Limiter modules (one for each band) for a mono system (or double those quantities for stereo). The flexibility of the SCAMP system allows additional functions to be added at any time to the basic system, such as S100 Gate or F300 Expander for single-ended noise reduction in any or all of the frequency bands; an SO7 Octave Equalizer for overall frequency shaping (particularly applicable to AM stations); an S06 Dynamic Noise Filter for the elimination of tape hiss; an S08 2 by 8 Distribution Amplifier for clean distribution of the audio signal around the station/studio, to or from the processor, and an S02 Transformerless Microphone Preamp, additional S01 Compressor, and S25 De-esser for all microphone/voice lines.

Mfr: Audio + Design Recording, Inc.

Circle 61 on Reader Service Card

12-CHANNEL PORTABLE MIXER

• The MX-P61 is a 12-channel multi-purpose portable mixer small enough to fit into a standard 19-inch rack, yet equipped with a large number of functions. The MX-P61 features all transformerless balanced input and output circuitry for optimal signal clarity. Each of its 12 input channels is equipped with Cannon-type connectors for either microphone or line inputs. There are four line outputs and three auxiliary outputs. Fully comprehensive monitoring and talkback sections are provided. The mixer also features three-band equalizers on each input, choice of VU metering or LED peak program meters, high- and low-cut filters, two stereo output limiters, and choice of AC or 12 volt DC power operation. The MX-P61 also offers selectable 48V or 12V AB phantom power and selectable line/auxiliary output reference levels of +4, +6, +8 dBm.

Mfr: Sony Professional Audio Products

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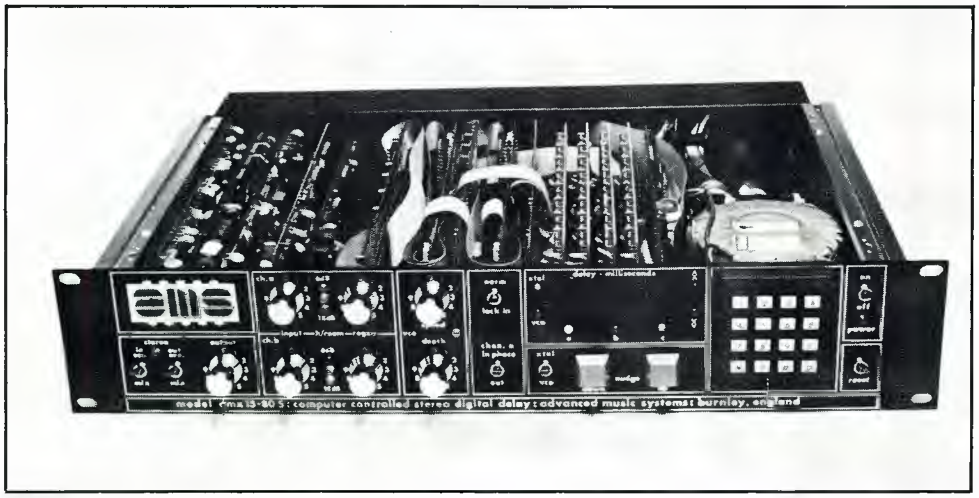
DIGITAL DELAY/PITCH CHANGER

• Advanced Music Systems' DMX 15-80S is a dual-channel modular system that can be expanded to give up to 32 seconds of delay at 18 kHz bandwidth and a 90 dB dynamic range. Features include keypad depth, phase reversal lock-in, and information storage by way of non-volatile memory. Two pitch changers with "de-glitch" module can be added, and software allows digital audio editing of locked information.

Mfr: Advanced Music Systems, Inc.

Circle 63 on Reader Service Card



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People, Places

• **Bernie Mann**, president of **Mann Media** (High Point, NC), was elected president of the **National Radio Broadcasters Association** October 2nd. Mann replaces **Sis Kaplan**, president of **WAYS/WROQ** Radio (Charlotte, NC), who had been president of NRBA for the past four years. Mann was elected unanimously by the 37 Board of Directors of NRBA at the recent radio convention in New Orleans. Bernie Mann has been on the Executive Committee as VP/East of the NRBA since 1979. He's also been elected Regional Director of NRBA for North and South Carolina since 1976. He owns radio stations **WGLD/WOKX** (High Point, NC) and **WKIK/WYYD** (Raleigh, NC). Mann has also been owner of **KALO/WEZQ** (Little Rock, AR) and General Manager/VP of **WROV** (Roanoke, VA) and **WAIR/WGPL** (Winston-Salem, NC). Before then, he worked as radio sales manager of **WAKE** (Atlanta), **WADO** (New York City) and **WTRY** (Troy, NY).

• **Hans D. Batschelet**, president of **Studer Revox America**, has announced the appointment of **Nancy M. Byers** as Eastern Regional Sales engineer and **Nick Balsamo** as Northeastern Regional manager. Working out of Studer's New York City office, Byers and Balsamo will assume primary responsibility for sales and service of Studer professional products throughout the Northeast, from Maine to Virginia. Ms. Byers comes to Studer from **CBS Recording Studios** in New York, where she served for 2½ years as a Staff Recording Engineer.

Before assuming his current post with Studer, **Nick Balsamo** served as National Sales manager for **Ikegami Electronics**. Balsamo began his career in audio/video technologies with a six year stint as a design engineer for **Harman Kardon**. Following two years with an engineering consulting firm, **Audio Dynamics Corporation**, Balsamo opened his own audio/video production facility, **Echo Sound Studios**. In 1974, after several years of consulting for **API**, Balsamo sold his interest in Echo and joined API full-time as manager of the Broadcast Products Division. From 1978 to 1982 he worked as an independent audio/video facilities design consultant. Balsamo is a member of **SMPTE**, **IEEE**, and the **AES**.

• **Dym/SR&A, Inc.**, a new agency providing public relations, promotion and marketing support services for high technology and electronics companies, has been established by **Fran Dym** and **Sumner Rider & Associates, Inc**. In addition to offering a full range of communications services to the trade, technical and consumer media and business press for companies in the audio, video, computer, telecommunications and electronic home security industries, Dym/SR&A will provide merchandising and promotion programs for such firms in conjunction with other clients in the shelter industry. Fran Dym, president of the new agency, was former senior vice president and Electronics/Technology Account Group head at **Daniel S. Roher, Inc.**, for such clients as **American Bell, Bang & Olufsen, BSR Canada, Ltd., dbx, Inc., Studer Revox America** and **Yamaha Electronics Co.** Headquarters of Dym/SR&A, Inc., will be at 355 Lexington Ave., New York, NY 10017.

• The **Time Capsule Brokerage**, a unique recording studio time brokerage and production coordination service, has begun its New York-based operations with a list of over 40 first-class studios and a client roster of over 200 repeat customers. Studio facilities and engineering personnel are available in and out of New York for a wide range of hourly or block rates. Services available include studio time bookings, instrument rental arrangements, engineer and musician contracting and budgetary consultation. Plans are in the works for expanding into A&R consultation services and other related areas. Typically, a prospective studio time customer calls the service to arrange for a booking. The client is then asked a series of questions concerning scheduling, budgetary, personnel, location, space, technical and musical requirements. Time Capsule then compiles a selection of facilities, personnel, scheduling and price that meet the stated requirements. After the client makes the appropriate choices, the brokerage takes over and coordinates all session-related activities.

• **Westrax Recording Studio**, located at 484 West 43rd Street, New York City, recently upgraded its 8 track facilities to 16 track. The new equipment includes a **Sound Workshop Series** 30 board, **Tascam** 85-16B 16 track with **dbx** Noise Reduction, **Lexicon** PCM-42 and a **Neumann** U-87 microphone. All equipment was purchased from **Martin Audio**, New York City. 2 and 8 track services continue to be available at Westrax. Recent additions to the studio include an OBXa **Oberheim** synthesizer and **LinnDrum** machine.

• **Peter Jensen** has been promoted to Midwest Regional sales supervisor for the Magnetic Tape Division of **Agfa-Gevaert, Inc.**, Teterboro, New Jersey. He will be based in the Oak Brook, Illinois Marketing and Training Center. Jensen, who is a member of the **Audio Engineering Society**, has been with Agfa-Gevaert for five and a half years. He was previously employed as Production Engineering manager with **International Audio, Inc.**

• **Digital Entertainment Corporation** (DEC) is pleased to announce the first major North American studio complex to purchase a complete digital audio

recording and editing system. **Lion Share Recording Studios** of Hollywood has just signed an agreement with DEC to purchase a **Mitsubishi Electric** Model X-800 32 channel digital audio recorder, two X-80 2 channel master recorders and the XE-1 digital audio electronic editing system. The multitrack and 2 track machines were scheduled for delivery around September 1st, with the XE-1 electronic editor to be delivered in November. Total contract value is in excess of $270,000. This major digital audio studio recording package is the first of any kind to be installed at a major studio. Lion Share will be in a unique position to supply first generation copies of master recordings for both analog and digital record and cassette mass production facilities. Lion Share Recording Studios is owned by **Kenny Rogers** and caters to many of the top recording artists in the world.

• **3M** announced that it has sold the service support capabilities and spare parts inventory for its professional analog audio recorders to **Electro-Technology Corporation**, Menlo Park, California. The sale includes a licensing agreement to manufacture spare parts to repair or rebuild the recorders which were last manufactured in 1979 by the former Mincom Products Division of 3M. According to **Art Cuscaden**, technical service supervisor, Broadcast and Related Products Division, the agreement includes all existing spare parts, engineering data, vendor information and test and manufacturing fixtures needed to provide repair services or parts to current owners of the equipment. In addition, the agreement provides for the training of Electro-Technology personnel in the use of the fixtures and equipment.

• **BTX** has appointed **James R. Lucas** manager of Western Area Sales, according to **Michael L. Sipsey**, vice president. Lucas will be responsible for all BTX sales and support activities in Arizona, California, Nevada, Oregon and Washington. Lucas has over eight years experience in the audio, video and film industries. He was most recently operating manager of **Command Video** in Los Angeles.

• **Sony Professional Audio Products** has named **Jeff Evans** and **Ernie De Los Santos** sales managers for the Western and Central regions, respectively. The announcement was made by **George Currie**, vice president and general manager. Each will have responsibility for **MCI/Sony** professional recording equipment, Sony wireless microphone systems and other Sony professional audio products. The appointments are part of an expansion of Sony's professional audio regional sales force designed to provide improved service and sales distribution to the recording, broadcast and sound reinforcement industries. Prior to joining Sony, **Evans** was a sales engineer for **Sound Genesis**, a San Francisco professional audio dealership. He has also served as project engineer for **Dolby Laboratories.** De Los Santos joined Sony Corporation of America in 1982 as Federal government marketing manager for Sony Video Communications. De Los Santos was previously a sales engineer for **Commercial Audio/Video Inc.**, a Houston sound reinforcement company.

• **Arthur C. Keller**, retired **Bell Labs** engineer who pioneered high fidelity and stereophonic recording techniques, died on August 25th at Lawrence Hospital in Bronxville, New York. He was 82.

Mr. Keller's invention of a "moving-coil" playback stylus made possible the first hi-fi records. His work with sound engineer **Irad S. Rafuse** led to the first single-groove stereophonic recordings, and their proposal for recording two sound channels onto a master disc became the standard stereophonic technique.

In 1931-32 Mr. Keller made the first known stereophonic and high-fidelity recordings of orchestral music during performances and rehearsals of the **Philadelphia Orchestra**, conducted by **Leopold Stokowski.** The recordings were part of a Bell Labs project aimed at improving the quality of recorded and amplified sound transmitted over the telephone network. In 1979 Mr. Keller identified and classified the Stokowski discs from among some 6,000 early recordings stored by Bell Labs at Murray Hill, New Jersey, and the following year Bell Labs presented a collection of the historic recordings to the nation's major record archives.

Mr. Keller was a Fellow of the **Acoustical Society of America** and the **Institute of Electrical and Electronic Engineers**, and a member of the **American Physical Society** and the **Yale Engineering Association.**

Audio Equipment Delivered to School of Audio Arts

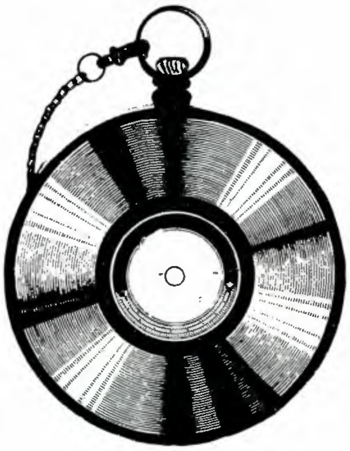
• The **School of Audio Arts**, the most recent addition to the expanding and recently relocated **Center for the Media Arts**, has accepted the final shipment of a large audio equipment package. The package consisted of an **Otari** MTR-90 Series II, 24 track, Otari MTR-10 2 track and 4 track recorders, 17 Otari MX5050B's, an Otari MX5050 MKIII-4, an **Eventide** SP2016 digital processor, **JBL** Bi-radial monitors, Ramsa 8118 and 8112 mixers, and a host of amplifiers, microphones and accessories.

THE HIGH C's TEST

• The final production of the **Greater Miami Opera**'s 1982-83 season, Verdi's opera, *The Masked Ball*, offered an opportunity to evaluate **B&K's** new microphones under the demanding and always fickle conditions of a live remote recording. **John Monforte** accompanied the microphones to the opera house where engineers **Ken Pohlmann** and **Carlos Santos** integrated them into their microphone array. A pair of coincident cardioids were removed and replaced with B&K spaced omnis, while **EV** shotgun microphones remained aimed into the deep stage. After several rehearsals and subsequent adjustments, three performances were taped, to be edited into a broadcast version for **National Public Radio**.

Pohlmann reported that the B&K omnis produced a clean and open sound, and monaural compatibility remained acceptable. He felt that the large diameter capsules were especially successful in providing a high output, low noise signal in this application. Following the final performance, tenor **Luciano Pavarotti** immortalized the box for the B&K microphones which had just recorded his voice by signing his name alongside the comment: "Thanks for a masterful recording."

db CD WATCH



• "The Compact Disc player is the fastest-selling new product in the history of audio technology," according to **John Briesch**, vice president, audio sales and marketing, **Sony Consumer Products Co.**, and a member of the **Compact Disc Group** (CDG) steering committee.

Briesch made his observation during a CDG presentation sponsored by the Los Angeles Chapter of the **National Academy of Recording Arts and Sciences.** The program, which introduced digital audio technology and Compact Disc players and recordings to more than 200 recording industry attendees, was run by the CDG, a non-profit association of 29 manufacturers of Compact Disc hardware and software supported by the **Recording Industry Association of America** (RIAA) and the **National Association of Recording Merchandisers** (NARM).

Briesch also noted that 18 manufacturers are or soon will be selling 22 CD player models in the U.S., including both Sony and **Denon**, who also offer professional units for radio station use. In addition, **Emiel Petrone**, a leading CDG spokesman and vice president marketing/Compact Disc coordinator—U.S.A., **PolyGram Records Inc.**, noted that, "by the end of this year, 260 classical, 177 pop, and 23 jazz titles will be on the U.S. market."

In related news, **Nippon Columbia**, Tokyo, Japan announced through its U.S. marketing arm, **Denon America Inc.** (DAI), the signing of a manufacturing agreement with **RCA** for the production of 1.5 million RCA Compact Discs over the next two years. The titles will be drawn from RCA's classical repertoire.

Sounds Like Mud (Island)

• The sound of music is better than ever at **Mud Island**, a Mississippi river theme park in Memphis, following a recent overhaul of the sound system at the island's 5,000 seat amphitheater.

The new sound package was provided by **Phase Audio, Inc.**, a Memphis-based sound contracting firm, with consultation from acoustician **Stephen Durr** of Nashville.

The purpose of the sound work was to enhance the design of the original system installed prior to Mud Island's opening more than a year ago. The original amplifiers and mixers were kept intact, but a major portion of the system, including speakers, horn drivers and equalizers, were replaced. The package also includes a new on-stage monitoring system.

The new speaker system, featuring a central cluster arrangement, includes a combination of **JBL** and **Gauss** speakers with a specialized crossover design incorporated into the system to achieve a flatter response. Phase Audio also installed **White** equalization for the main speakers and on-stage monitors.

The new amphitheater sound system at Mud Island, featuring speakers in a central cluster arrangement above the stage.

Ahoy There, Sailor

Now comes Miller time.

• From the "how I spent my summer vacation" file. **Lexicon**'s chief executive officer, **Ron Noonan**, not only runs one of the world's leading professional audio and broadcast equipment companies, he also has joined a select group of captains who have won the coveted Marion (USA to Bermuda) cruising race. Sailing in G class with his daughter, Michelle, son, Rick, and crew, Noonan's Bristol 40 sloop Wildflower won overall first place competing against 134 yachts in the 645 nautical mile race. Corrected time was 3 days, 17 hours, 47 minutes and 18 seconds.

Rick Wakeman (center), currently working on the music score for SHE, *pictured with studio engineers Peter Wandless (left) and Dick Lewzey (right) and the new Sony 3324 and 1610 digital recorders.*

CTS Goes Digital

• **CTS Studios Limited** have completed the first stage of their digital overhaul with the acquisition of the **Sony** PCM 3324 digital 24 track recorder and the Sony PCM 1610 digital stereo recording and editing system at their Wembley studios. CTS is the first U.K. studio to offer the Sony 24 track digital state-of-the-art recorder permanently in-house. The second stage of the digital overhaul will take place in early 1984 when the **Neve** DSP all-digital console is installed in CTS Studio 1. CTS will then have the world's first comprehensive all-digital studio.